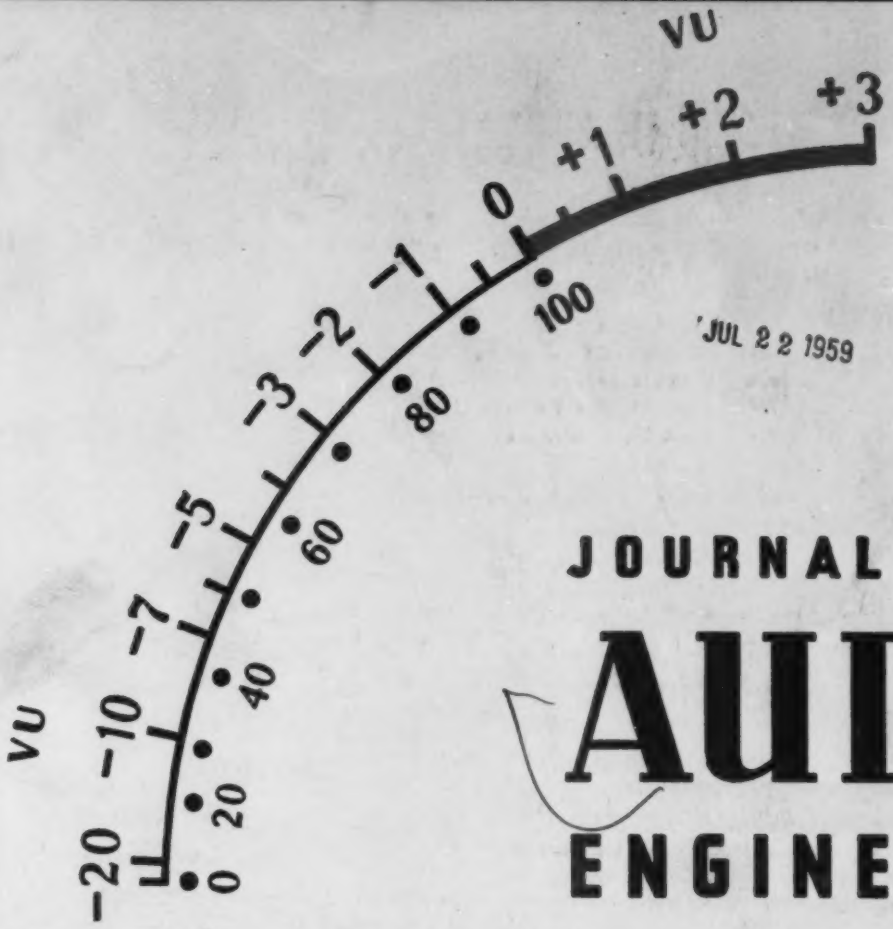

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JOURNAL OF THE AUDIO ENGINEERING SOCIETY

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VOLUME 7 NUMBER 2 APRIL 1959

ELEVENTH ANNUAL
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Morning	Transistors and Audio Circuit Applications	Paul A. Grace, Raytheon Manufacturing Company
Afternoon	Artificial Reverberation	John M. Hollywood, CBS Laboratories
Evening	Music and Electronics	Vladimir Ussachevsky, Columbia University

Tuesday, October 6

Morning	Audio Applications	S. Edward Sorensen, Columbia Records, Inc.
Afternoon and Evening	Disc Recording and Reproduction	Benjamin B. Bauer, CBS Laboratories

Wednesday, October 7

Morning	Amplifiers	Rein Narma, Fairchild Recording Equipment Corp.
Afternoon and Evening	Magnetic Recording and Reproduction	Walter H. Erikson, Radio Corporation of America

Thursday, October 8

Morning	Audio Measurements and Standards	Sheldon Wilpon New York Naval Shipyard
Afternoon	Stereo Perception	R. C. Moyer, Record Division, RCA Victor

Friday, October 9

Morning	Studio and Speech Input Systems	Philip C. Erhorn, Audiofax, Inc.
Afternoon and Evening	Loudspeakers	Abraham B. Cohen, Advanced Acoustics Company

Journal of the AUDIO ENGINEERING SOCIETY

APRIL 1959 VOLUME 7, NUMBER 2

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How Much Audio Power?*

STANLEY H. SESSIONS

U. S. Navy Electronics Laboratory, San Diego 52, California

Some current estimates of audio power amplifier requirements for home listening systems are exaggerated. A consideration of preferred listening levels, the acoustic characteristics of rooms, the nature of reproduced sound, especially music, and peak audio power lead the author to conclude that an accurately rated 10-watt amplifier is sufficient for most music listening requirements.

INTRODUCTION

NOWHERE IN audio technology is there as much disagreement as over the question of the amount of audio amplifier output power needed for listening systems. Estimates of audio power required for high quality home systems, for example, range from a few microwatts to kilowatts. Is it any wonder that the newcomer to audio is confused when engineers themselves disagree so widely?

This paper does not purport to settle the question of how much audio power is needed. It attempts to indicate some of the factors which determine audio power requirements in the most typical room enclosures. It may also encourage engineers to investigate the matter of audio power in greater detail, thereby helping all who work in this rapidly expanding field.

LISTENING LEVELS

A common and often quoted misconception is that pleasant listening requires the loudspeaker to reproduce the same level of sound as produced by the original source. Such levels would actually be disagreeable if not intolerable in most listening rooms. This condition in small rooms occurs largely because of the contribution of reflected sound to the total sound intensity in the room. Even beautiful music is not pleasurable when amplified to high intensity levels.

Knudsen,[1] in describing the design of rooms for live music performance, illustrates the matter of preferred music listening levels by indicating the size of room which is suitable for various kinds of musical performance. He states that the most preferred listening level is at an average maximum intensity of about 68 db. In a room having a volume of 100 m^3 and a reverberation time of 1.0″, this corresponds to the output of two average singers or approximately 200 μw of acoustic power. At times, of course, the level may increase a hundred-fold. In contrast, a room suitable for an orchestra of 100 would require a volume of about 30 000 m^3, about equivalent to a concert hall seating 5 000 with a 2-sec reverberation time.

A study of preferred listening levels was made by Chinn and Eisenberg[2] in 1947. Results of this study showed that the average listener preferred average peak program levels of 65 to 70 db. A later study by Somerville and Brownless[3] for the B.B.C. using both live and recorded broadcasts has indicated that preferred levels might be somewhat higher. Tests were extended to include listeners' level preferences both in studios and in their own living rooms. These tests agreed remarkably well. The preferred *maximum* sound levels found are shown in Table I. It was also found that the level preference decreases significantly with age.

TABLE 1. Preferred maximum sound level (db *re* 10^{-16} watts/cm^2).

	Public		Musicians
	Men	Women	
Symphonic music	78	78	88
Light music	75	74	79
Dance music	75	73	79
Speech	71	71	74

In another listener sampling, the preferences of broadcast transmission engineers were measured. Here it was found that the level preference of engineers was significantly higher than that of average radio listeners. This is no doubt due to the habits of engineers in listening for quality differences in program materials. The latter level preferences might be similar to those of high fidelity music listeners.

* Delivered at the West Coast Convention of the Audio Engineering Society, Los Angeles, February 8, 1957. Revised manuscript received September 23, 1957.

[1] V. O. Knudsen, "Acoustics of Music Rooms," *J. Acoust. Soc. Am.*, **2**, 439 (1939).

[2] H. A. Chinn and Philip Eisenberg, "New CBS Program Transmission Standards," *Proc. IRE*, **35**, 1547 (1947).

[3] T. Somerville and S. F. Brownless, "Listener's Sound-Level Preferences," *BBC Quarterly*, **3**, 4, 245 (1949).

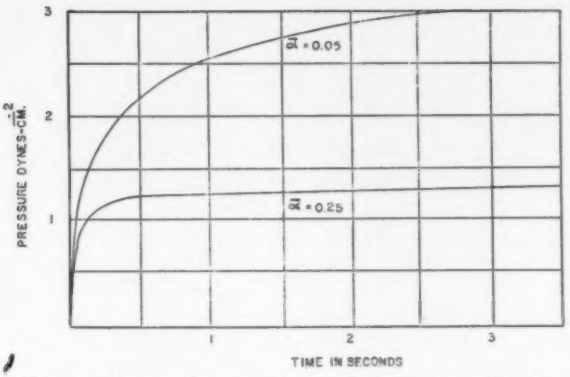


FIG. 1. The growth of sound as a function of reverberation. (After Knudsen and Harris.)

Room Effect on Sound Intensity

Because the walls of a room act as sound reflecting surfaces, the intensity of a generated sound is, at any time, due to both direct and reverberant energy. Since music contains varying amounts of short and long duration tones, its sound level is produced by the total of both direct and persisting reverberant sound at any instant of time.

Sound pressure from a single tone source in a room increases as a function of time. This growth in sound pressure in a room has been described by Knudsen and Harris[4] who point out that the time for sound to reach a maximum pressure can be related to the reverberation time of the room. Pressure versus time curves for two values of room reverberation are shown in Fig. 1. This figure indicates that maximum pressure in a room having a 0.25 second reverberation time is reached in approximately 0.2 second. The corresponding sound field dies out at a much slower rate. In the case of reproduced music a large part of the sound pressure is made up of reverberant energy continuing after the initial sound formation.

Knudsen and Harris illustrate this growth of sound pressure as a function of time, (Fig. 1). The higher coefficient ($\bar{a} = 0.25$) might be typical of the well-furnished living room. A simplified expression for the steady state pressure in a room is given by,

$$P = 1300\ (W/a)^{1/2}$$

in which P is the sound pressure in dyne/cm^2, W is the acoustic power of the source in watts, and a is the number of absorption units in sabines. This formula yields an approximate value of steady state pressure in a room at distances greater than about 5 feet from the source.

[4] V. O. Knudsen and C. M. Harris, *Acoustical Designing in Architecture,* (John Wiley and Sons, New York, 1950), pp. 145-147.

Nature of Reproduced Sound

It is felt by the writer that while the steady state calculations for room intensity do not completely express average sound pressures in a room, they provide useful averages when some margins are allowed for type of sound and angle of sound distribution.

In view, also, of the present techniques of "live" microphone pickup added reverberation in the original microphone pickup tends to cause more uniform distribution of sound energy as seen by the microphone. If, for example, microphone pickups of music were made at distances 10 to 20 feet from the orchestras, this would, in rooms having reverberation times in excess of 1 second, tend to excite the rooms to produce their sustained pressure levels.

As a check of the nature of music envelopes, continuous trace oscillograph recordings have been made of a variety of types of concert band music as recorded in a studio having a reverberation time of about 1 second. Excerpts of these are shown in Fig. 2. The film speed used was 10 ips.

Since most music employs sustained tones except between phrases or at times of percussive attack, it seems reasonable to use average sound pressure and steady state pressure values interchangeably for vocal and orchestral music. That this assumption is not precise or applicable to all music forms is readily apparent, but for the major part of reproduced music steady state formulas seem applicable.

Audio Peak Power

The audio power requirements to accommodate peak pressure levels seem frequently to have been exaggerated. In the very excellent study of Sivian, Dunn, and White[5] the greatest instantaneous to average sound power ratios meas-

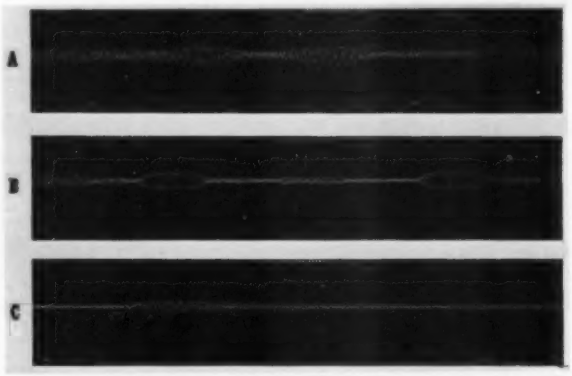


FIG. 2. Oscillograms made from recordings of concert band. A. Band music with cymbal crashes, peak-to-average power approximately 12 db. B. Band music with clarinet solo, peak-to-average power approximately 9 db. C. Band music with bell solo, peak-to-average power approximately 6 db.

[5] L. J. Sivian, H. K. Dunn, and S. D. White, *J. Acoust. Soc. Am.*, 2, 330 (1931).

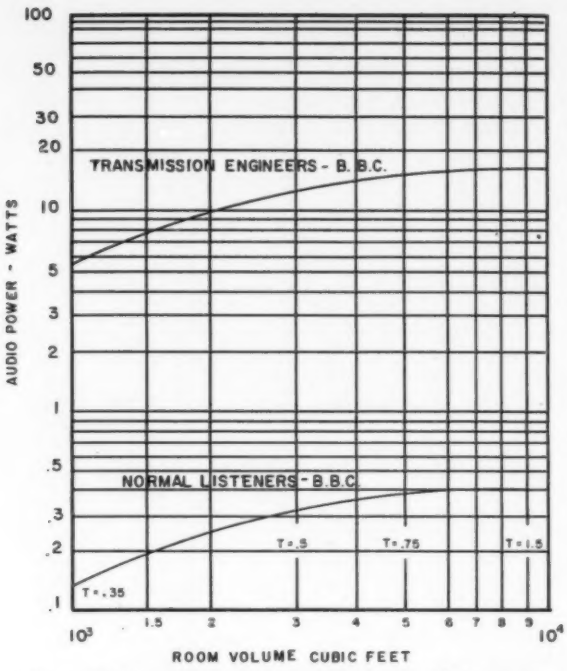


FIG. 3. Audio amplifier power required in typical living rooms.

ured were approximately 30 db. This peak was reached with a large bass drum and does not seem typical. A more realistic maximum peak based on the majority of the Bell Laboratories measurements and those just discussed seem to indicate that a 20 db ratio between maximum average power and peak power would be ample. In the oscillographic records made of the concert band no peak-to-average ratios were found exceeding 12 db.

No attempt has been made here to relate the term "peak" power, as discussed in amplifier power output ratings, with sine wave output power. It has been assumed that an amplifier should be capable of supplying power on the basis of its sine wave rating as a safety factor in making power estimates.

Power Estimates

On the basis of the steady state acoustic power requirements of small rooms a calculation has been made for the audio power required for orchestral music reproduction at levels indicated in the B.B.C. survey. The results of these computations are plotted in Fig. 3. They have been made for a loudspeaker having 5 percent efficiency. The curve has been calculated for the average absorption found in several typical living rooms by Gales and Young.[6] An allowance of 20 db has been made for peak power requirements.

These curves should be helpful in estimating the power needs for listening to reproduced sound in typical living rooms. Attention should be given to the loudspeaker efficiency (compared to 5 per cent). Total room absorption and the listener level preference will of course affect the audio power required.

It is interesting to note that the calculated values for audio power required fall into a range agreeing well with current audio practices. In group demonstrations in which a 10-watt amplifier was compared with a 50-watt amplifier in a 500-seat auditorium at somewhat higher than preferred listening levels, the most significant comment was that the 10-watt amplifier had more distortion at all levels than the 50-watt. This observation would seem to point up the acute awareness of the ear to small percentages of distortion. It is also fortunate, no doubt, that the transient behavior of amplifiers is being subjected to careful scrutiny by engineers. A rather well-known weakness of lower-powered amplifiers is their failure to supply rated power at low frequencies. This, of course, must be accounted for in audio power estimates since some of the highest power-producing components occur at low audio frequencies.

CONCLUSION

Perhaps there is a tenable middle ground between the microwatt and kilowatt amplifier proponents. Estimates based on listener preference tests and steady state power levels in small rooms seem to coincide with limited experimental evidence. Refined measurements may give more precise figures for establishing the audio power requirements of the typical listener.

It appears that audio power requirements for the average listener have often been grossly overestimated. By estimating room sizes, absorption, loudspeaker efficiency, and listening level preferences a more realistic choice between maximum and minimum power can be made. An accurately rated 10-watt amplifier would serve most music listening requirements. Such a choice must however be based on honest amplifier ratings.

[6] R. S. Gales and R. W. Young, "Some Acoustical Considerations in Listening to Reproduced Music in the Home," (unpublished).

The Distribution of Peak Energy in Recorded Music, and Its Relation to Magnetic Recording Systems*

JOHN G. McKNIGHT†

Ampex Corporation, Redwood City, California

This is the author's second paper on this subject. The abstract from the first paper is reprinted from January, 1959, issue of the JOURNAL OF THE AUDIO ENGINEERING SOCIETY.

Measuring procedures and specifications currently used in the sound recording industry are inadequate. A weighting network should be used in objective noise measurement work to evaluate developments toward a lower noise system.

We know that the ear is most sensitive in the 1 to 6 kc region, and it is possible to use preemphasis in this region. Better signal-to-noise ratio may be obtained in 15 ips magnetic recorders by utilizing fully their present capabilities. Subjective listening tests show that a system with the 1 to 6 kc region pre-emphasized (the AME curve) is some 7 db quieter than the same system using the NAB curve, but does not show any audible increase in distortion.

Data is needed on the distribution of peak energy in music in order to design preemphasis on the basis of equal probability of overload at all frequencies. Previously published data is discussed, and new data is presented and discussed. The data show that energy distribution is very variable; the energy may be within plus or minus 3 db from 40 c to 16 kc; or the energy may be down as much as 30 db at 40 c and 6 kc.

INTRODUCTION

THERE ARE TWO limiting criteria for designing equalization in a recording system. One criterion is that of making a post-emphasis which will minimize the audible noise from the system, using at the same time a volume indicator to insure that the recording system will not be overloaded.[1]

The second criterion is that of designing the pre-emphasis so that the system will have equal probability of overload at all frequencies; then the volume indicator may be designed primarily for *balancing* levels.

In order to evaluate equalizations on the basis of this second criterion of equal probability of overload at all frequencies, it is necessary to have adequate data on the frequency-distribution of peak energy. In the present case we are interested in music only.

Previous work was reviewed and found to be inadequate for our purposes;[2] therefore, we undertook our own measurements of the distribution of peak energy in recorded music. The apparatus and recordings are described here, and a few of the spectrum analyses presented.

REVIEW OF PREVIOUS STUDIES

A review of the definitive Sivian, Dunn and White analyses is very interesting.

Fig. 1 shows a summary of their spectrum analyses of a 75 piece orchestra in a theatre. The heavy curve shows the only spectrum reprinted in Olson's *Acoustical Engineering*;[3] the implication is that this is a typical spectrum; note, however, that the other spectra have much greater high-frequency content. These upper spectra are similar to some of the data in this new study, and show that the general engineering interpretation of Sivian, Dunn, and White's data (as selected by Olson) may not be correct.

Several problems arise in interpreting the Sivian, Dunn and White data. First, their data is unrelated to VU meter readings, having been done long before the development of

* Received April 20, 1959. Presented February 20, 1959, at the Sixth Annual Western Convention of the Audio Engineering Society, Los Angeles, Calif.

† Senior Engineer, Research Division. Now Manager, Advanced Audio Section, Professional Audio Department.

1. J. G. McKnight, "Signal-to-Noise Problems, and a New Equalization for Magnetic Recording", *J. Audio Eng. Soc.*, 7: 5-12 (Jan. 1959)

2. Sivian, Dunn, and White "Absolute Amplitudes & Spectra of Certain Musical Instruments and Orchestras," *J. Acous. Soc. Amer.* 2: 330-371 (January 1931).

3. H. F. Olson, *Acoustical Engineering*, D. Van Nostrand Co., New York, 1957, p 589, Fig. 12. 35.

the VU meter. Second, their spectra are for one orchestra in one pick-up situation, playing one type of musical arrangement. This is a very small sample on which to base conclusions about all recordings of all music. Also, we would like to know the spectrum of the signal from a commercial recording session, where there may be many microphones placed close to the individual instruments, individual microphone equalization, and other techniques for "better-than-being-there" sound.

Analyses were done in 1953 at Ampex Corporation by George Brettell.[4] Fig. 2 shows a summary of his spectra. Again, considerable high-frequency energy is shown, but too few samples are here to draw any general conclusions.

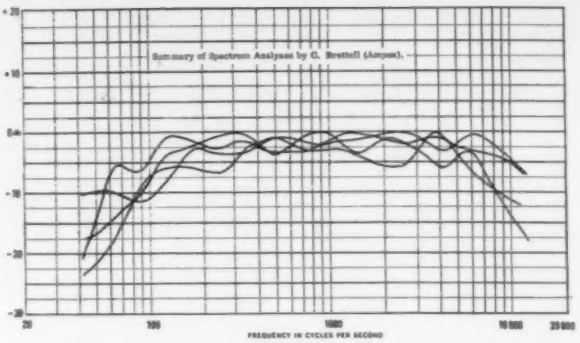


FIG. 2.

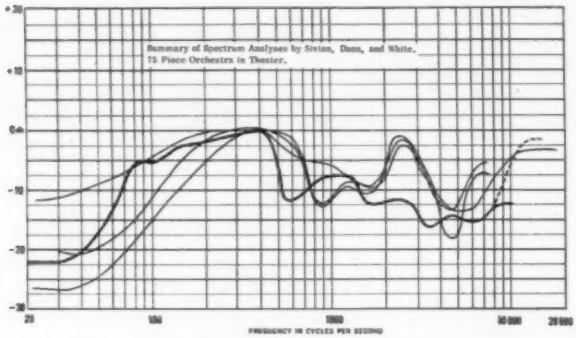


FIG. 1.

We question the data in J. P. Overley's article, "Energy Distribution in Music,"[5] on several accounts. Although he describes the need for peak readings, his instruments would read only average amplitudes. The data was taken from phonograph pressings. And the published data is averaged from many readings. We will see later in the data of this paper that "averages" have little meaning—we are concerned entirely with individuals. The Overley data shows much less energy above 3 kc than any of the other studies discussed in this paper.

Our review has shown us, then, that the general engineering conception of energy falling at high frequencies does not follow from the Sivian, Dunn, and White data; both their and the Brettell spectra show energy beyond 10 kc with amplitude within 3 to 5 db of the mid-band energy.

THE NEW SPECTRUM ANALYSES

The need was now for a large number of spectrum analyses to show the variation and limits of spectra from recordings made of many different types of groups, in many different recording situations, with many different kinds of musical arrangements being played.

First, it was necessary to design and construct equipment which would take the spectrum analysis data accurately and easily. This required a memory device, so that the recording to be analyzed could be played one time, and the complete spectrum read. Fig. 3 shows the block diagram of the audio spectrum analyzing system. The tape reproducer and the line amplifier feed both an attenuator and VU meter; and also through a transformer into a set of filters, which divide the audio spectrum (40 c through 16 kc) into 27 bands, each one-third octave wide. The 27 filters plus one unfiltered channel feed a 28-channel "maximum peak" memory unit, to be described.

In operation, the memory unit was first cancelled; the program to be analyzed was played, during which time the VU attenuator was adjusted. Then the spectrum, the total peak level, and the VU attenuator (giving VU level) were read—and that spectrum analysis was completed.

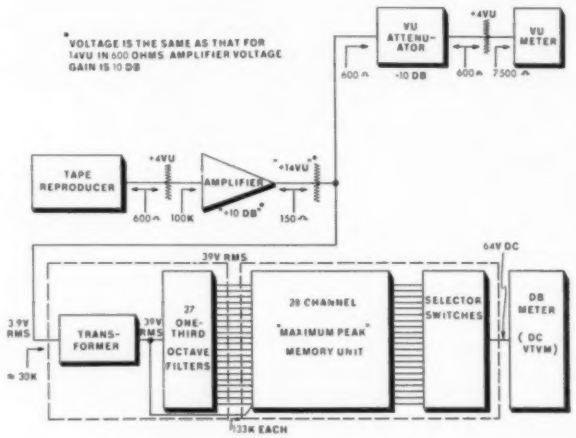


FIG. 3.

4. G. Brettell, Ampex Corp. unpublished engineering data (1953).
5. J. P. Overley, "Energy Distribution in Music", *IRE, Trans. on Audio*, **5**: 120-123 (September-October 1956).

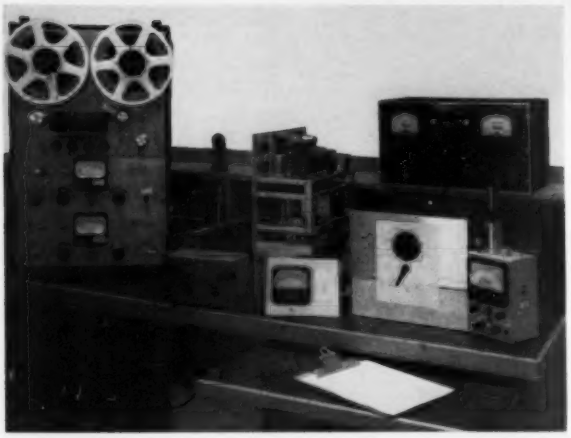


FIG. 4.

Fig. 4 is a photograph of the equipment: tape reproducer, line amplifier, VU attenuator and meter, filter, memory unit and its power supply, and the VTVM used as a db meter from the memory.

The individual "maximum peak" memory units are simple and straight-forward. Fig. 5 shows a typical channel. An input level calibrate control feeds a pentode amplifier which is followed by a cathode follower, which, in conjunction with the 3A2 rectifier, charges the capacitor. The time constant of the charging circuit is such as to charge the capacitor to within 1 db of true peak amplitude in about 1 ms. Following peaks of smaller amplitude have no effect; larger ones charge the capacitor to the new peak value, regardless of previous charge. The back resistance of the 3A2, in parallel with the input resistance of the meter used for readout, is such as to discharge the capacitor about 1 db in 10 minutes.

Fig. 6 shows the memory unit (clip-lead connected to the one-third octave filter), the output meter, and the memory unit power supply. The input level calibrate controls, pentode-triode amplifiers, rectifiers, capacitors, and selector switches are visible on top of the chassis.

The material to be analyzed was chosen from what we will call "original recordings" and "studio recordings." We recorded most of the "original recordings" with a single condenser microphone, so placed as to make what we considered a good sounding recording. No special "better-than-being-there" techniques were used, and we feel the recordings are representative of the sounds actually made by the instruments.

The "studio recordings" are tapes made at the various major U.S. studios, and represent what may be fed into tape recorders, including the effects of multiple close microphones with individual equalization, echo chambers, etc.

We made analyses of "original recordings" of 34 different sections from 18 entirely different recording situations; and of "studio recordings" of 28 samples from 20 different situations.

We will show 15 analyses here, selected to show the *variation* encountered in "original" and "studio" recordings, for symphony orchestra, chamber music, pipe organ, dance, jazz, and popular music. We will also show a summary of the peak factors, and a summary of the limits of the spectra encountered. We feel that the concept of an "average" spectrum is not very useful. If it rains today, you'll get wet even if the "average" weather for today is supposed to be "dry"!

The discussion will be directed toward the high frequencies, but similar variation is found in the low frequency energy.

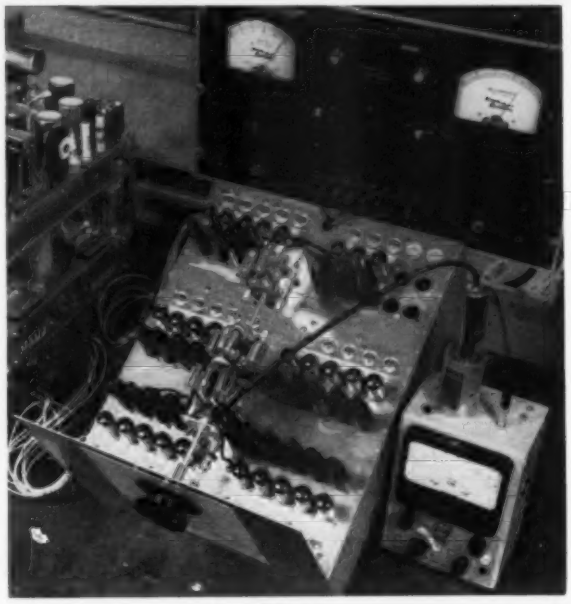


FIG. 6.

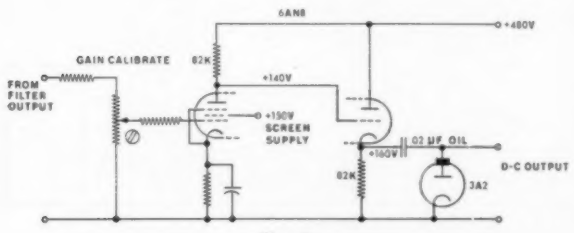


FIG. 5.

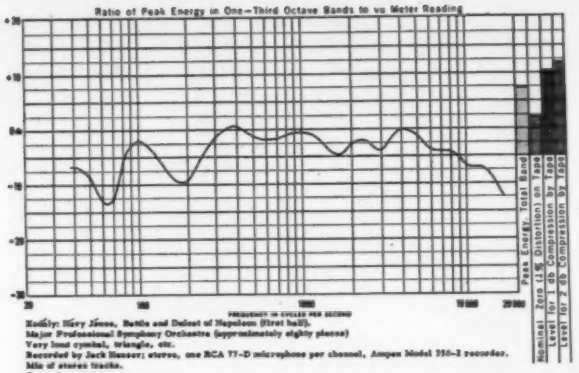
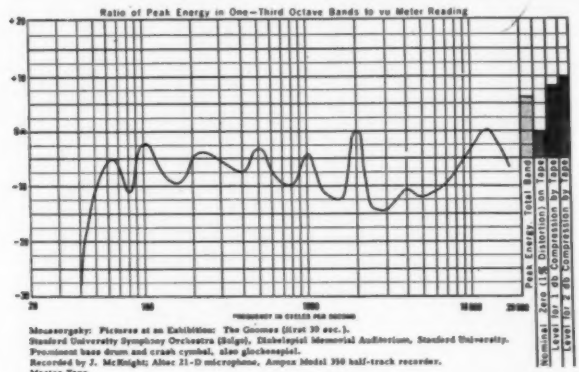

Ratio of Peak Energy in One-Third Octave Bands to vu Meter Reading
Peak Energy, Total Band
Nominal Zero (1% Distortion) on Tape
Level for 1 db Compression by Tape
Level for 2 db Compression by Tape
FREQUENCY IN CYCLES PER SECOND
Kodály: Háry János, Battle and Defeat of Napoleon (first half).
Major Professional Symphony Orchestra (approximately eighty pieces)
Very loud cymbal, triangle, etc.
Recorded by Jack Hauser; stereo, one RCA 77-D microphone per channel, Ampex Model 350-2 recorder.
Mix of stereo tracks.
Copy of master tape.

FIG. 7.

Figs. 7, 8, and 9 show three analyses of full symphony orchestras—original recordings of a professional and a university orchestra, and a studio recording of a professional


Ratio of Peak Energy in One-Third Octave Bands to vu Meter Reading
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FREQUENCY IN CYCLES PER SECOND
Mussorgsky: Pictures at an Exhibition: The Gnomes (first 30 sec.).
Stanford University Symphony Orchestra (Solgo), Skobeloptsi Memorial Auditorium, Stanford University.
Prominent bass drum and crash cymbal, also glockenspiel.
Recorded by J. McKnight; Altec 21-D microphone, Ampex Model 350 half-track recorder.
Master Tape.

FIG. 8.

orchestra. These were chosen to show a large amount of high frequency energy. (The curves are all plotted relative

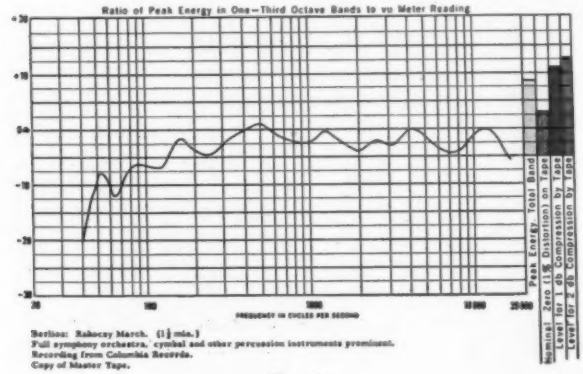
Ratio of Peak Energy in One-Third Octave Bands to vu Meter Reading
Peak Energy, Total Band
Nominal Zero (1% Distortion) on Tape
Level for 1 db Compression by Tape
Level for 2 db Compression by Tape
FREQUENCY IN CYCLES PER SECOND
Berlioz: Rakoczy March. (1½ min.)
Full symphony orchestra, cymbal and other percussion instruments prominent.
Recording from Columbia Records.
Copy of Master Tape.

FIG. 9.

to the VU meter reading for each selection as "zero." The first bar at the right of each graph shows the peak energy in the total band. The others show where the 1% total harmonic distortion level was, and where 1 and 2 db of compression in mid-band by the tape would occur. These will be discussed in a future paper.)

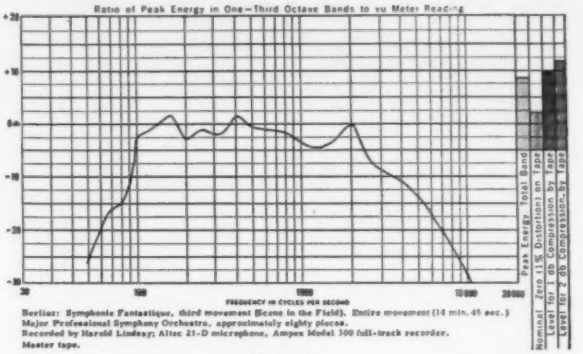
Ratio of Peak Energy in One-Third Octave Bands to vu Meter Reading
Peak Energy, Total Band
Nominal Zero (1% Distortion) on Tape
Level for 1 db Compression by Tape
Level for 2 db Compression by Tape
FREQUENCY IN CYCLES PER SECOND
Berlioz: Symphonie Fantastique, third movement (Scene in the Field). Entire movement (14 min. 45 sec.)
Major Professional Symphony Orchestra, approximately eighty pieces.
Recorded by Harold Lindsay; Altec 21-D microphone, Ampex Model 300 full-track recorder.
Master tape.

FIG. 10.

On the other hand, Figs. 10 and 11 show analyses of professional full symphony orchestras. Here we see very little high-frequency energy.

Figs. 12 and 13 show two original recordings of chamber music; Fig. 12, with a tambourine, shows appreciable high-frequency energy; Fig. 13, a baroque orchestra of strings with some winds, shows relatively little high-frequency energy.

Fig. 14 shows a church pipe organ. Note the large amount

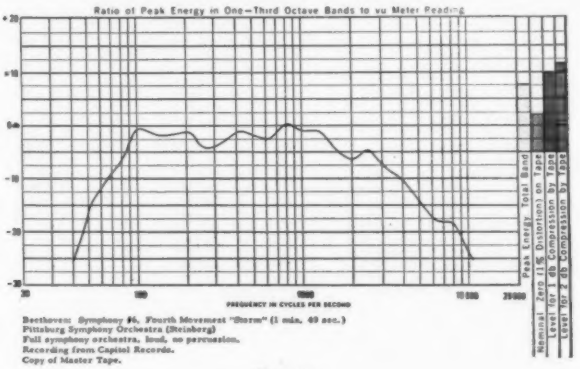
Ratio of Peak Energy in One-Third Octave Bands to vu Meter Reading
Peak Energy, Total Band
Nominal Zero (1% Distortion) on Tape
Level for 1 db Compression by Tape
Level for 2 db Compression by Tape
FREQUENCY IN CYCLES PER SECOND
Beethoven: Symphony #6, Fourth Movement "Storm" (1 min. 40 sec.)
Pittsburg Symphony Orchestra (Steinberg)
Full symphony orchestra, loud, no percussion.
Recording from Capitol Records.
Copy of Master Tape.

FIG. 11.

of low-frequency energy, with a steady slope downward toward higher frequencies. Recordings of a theater pipe organ, despite the percussion and other effects, had similar spectra.

Figs. 15 and 16 show original recordings of a large pro-

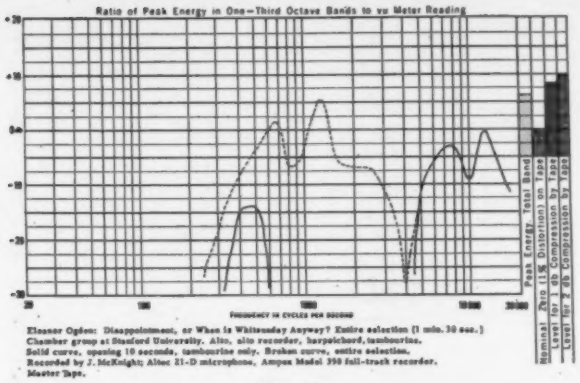
Ratio of Peak Energy in One-Third Octave Bands to vu Meter Reading
FREQUENCY IN CYCLES PER SECOND
Peak Energy, Total Band
Nominal Zero (1% Distortion) on Tape
Level for 1 db Compression by Tape
Level for 2 db Compression by Tape
Eleanor Ogden: Disappointment, or When Is Whitsunday Anyway? Entire selection (1 min. 30 sec.)
Chamber group at Stanford University. Alto, alto recorder, harpsichord, tambourine.
Solid curve, opening 10 seconds, tambourine only. Broken curve, entire selection.
Recorded by J. McKnight; Altec 21-D microphone, Ampex Model 350 full-track recorder.
Master Tape.

FIG. 12.

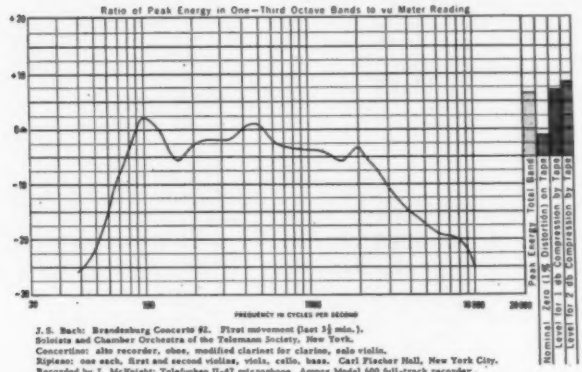
Ratio of Peak Energy in One-Third Octave Bands to vu Meter Reading
FREQUENCY IN CYCLES PER SECOND
Peak Energy, Total Band
Nominal Zero (1% Distortion) on Tape
Level for 1 db Compression by Tape
Level for 2 db Compression by Tape
J. S. Bach: Brandenburg Concerto #2. First movement (last 3½ min.).
Soloists and Chamber Orchestra of the Telemann Society, New York.
Concertino: alto recorder, oboe, modified clarinet for clarino, solo violin.
Ripieno: one each, first and second violins, viola, cello, bass. Carl Fischer Hall, New York City.
Recorded by J. McKnight; Telefunken U-47 microphone, Ampex Model 600 full-track recorder.
Copy of the 7½ ips Master Tape.

FIG. 13.

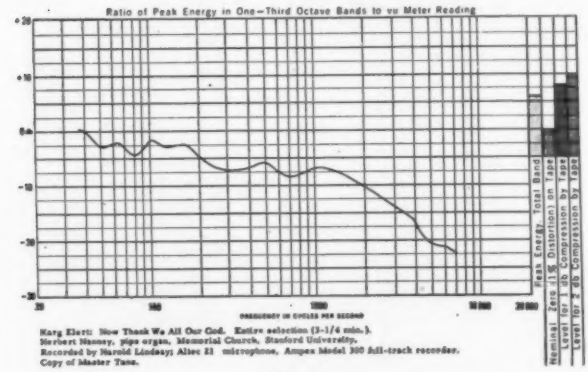
Ratio of Peak Energy in One-Third Octave Bands to vu Meter Reading
FREQUENCY IN CYCLES PER SECOND
Peak Energy, Total Band
Nominal Zero (1% Distortion) on Tape
Level for 1 db Compression by Tape
Level for 2 db Compression by Tape
Karg Elert: Now Thank We All Our God. Entire selection (3-1/4 min.).
Norbert Nanney, pipe organ, Memorial Church, Stanford University.
Recorded by Harold Lindsay; Altec 21 microphone, Ampex Model 300 full-track recorder.
Copy of Master Tape.

FIG. 14.

fessional jazz band in concert (including cow bells and all!) and a small non-professional dance band. Fig. 15 shows a medium amount of high-frequency energy, Fig. 16, less high-frequency energy.

Fig. 17 and 18 show two studio recordings of professional dance bands. Fig. 17 shows a strong peak in the 5 kc region, from a muted trumpet solo. Fig. 18 is of a dance band with solos by piano, muted trumpets, saxophones and strings. The spectrum is practically flat to 10 kc.

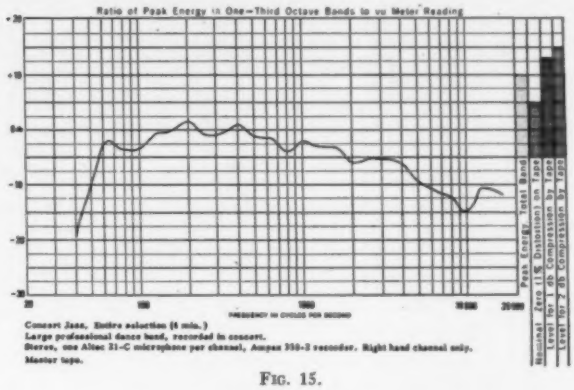
Ratio of Peak Energy in One-Third Octave Bands to vu Meter Reading
FREQUENCY IN CYCLES PER SECOND
Peak Energy, Total Band
Nominal Zero (1% Distortion) on Tape
Level for 1 db Compression by Tape
Level for 2 db Compression by Tape
Concert Jazz. Entire selection (6 min.)
Large professional dance band, recorded in concert.
Stereo, one Altec 21-C microphone per channel, Ampex 350-2 recorder. Right hand channel only.
Master tape.

FIG. 15.

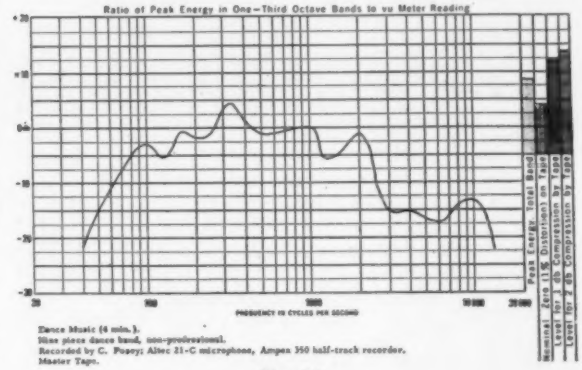
Ratio of Peak Energy in One-Third Octave Bands to vu Meter Reading
FREQUENCY IN CYCLES PER SECOND
Peak Energy, Total Band
Nominal Zero (1% Distortion) on Tape
Level for 1 db Compression by Tape
Level for 2 db Compression by Tape
Dance Music (4 min.).
Nine piece dance band, non-professional.
Recorded by C. Pusey; Altec 21-C microphone, Ampex 350 half-track recorder.
Master Tape.

FIG. 16.

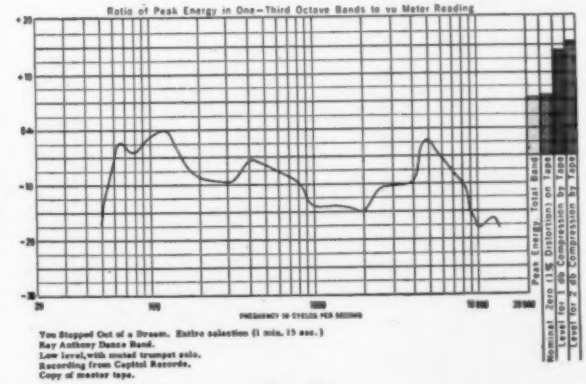
Ratio of Peak Energy in One-Third Octave Bands to vu Meter Reading
FREQUENCY IN CYCLES PER SECOND
Peak Energy, Total Band
Nominal Zero (1% Distortion) on Tape
Level for 1 db Compression by Tape
Level for 2 db Compression by Tape
You Stepped Out of a Dream. Entire selection (3 min. 15 sec.)
Ray Anthony Dance Band.
Low level, with muted trumpet solo.
Recording from Capitol Records.
Copy of master tape.

FIG. 17.

Figs. 19 and 20 show the two different tracks of a stereo recording (neither track contained any sound from the other) of a studio master tape of the Perez Prado band, playing "Patricia." The melody instruments channel (Fig. 19)

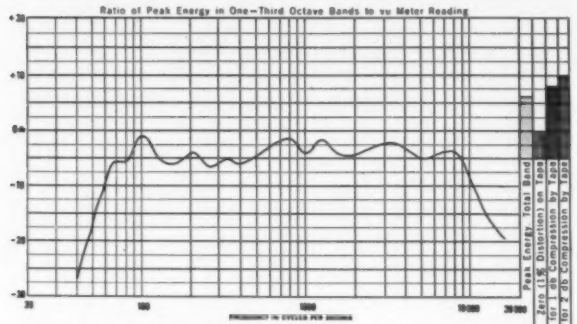
Ratio of Peak Energy in One-Third Octave Bands to vu Meter Reading
Peak Energy, Total Band
Nominal Zero (1% Distortion) on Tape
Level for 4 db Compression by Tape
Level for 2 db Compression by Tape
Dance Music (title unknown). Entire selection (2½ min.)
Dance band with solo by piano, muted trumpets, saxophones and strings.
Recording from W. S. Miltenberg, RCA: stereo, right hand channel.
Master Tape.

FIG. 18.

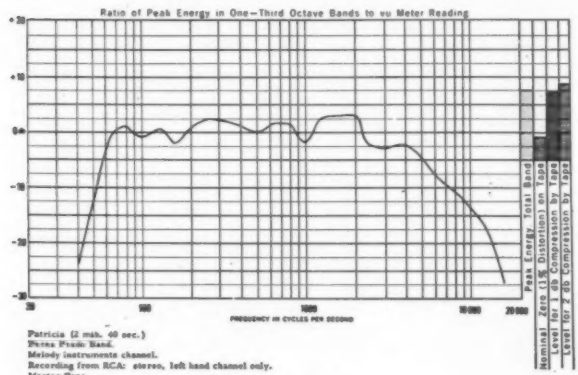
Ratio of Peak Energy in One-Third Octave Bands to vu Meter Reading
Peak Energy, Total Band
Nominal Zero (1% Distortion) on Tape
Level for 4 db Compression by Tape
Level for 2 db Compression by Tape
FREQUENCY IN CYCLES PER SECOND
Patricia (2 min. 40 sec.)
Perez Prado Band.
Melody instruments channel.
Recording from RCA: stereo, left hand channel only.
Master Tape.

FIG. 19.

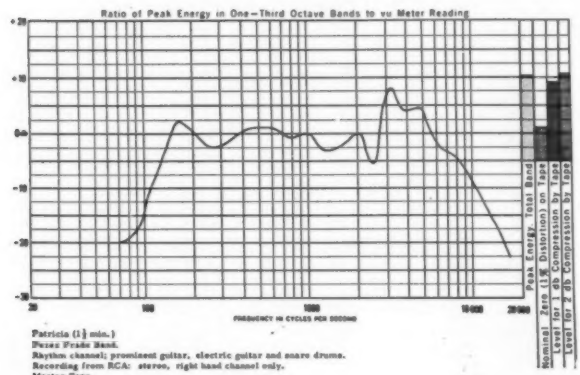
Ratio of Peak Energy in One-Third Octave Bands to vu Meter Reading
Peak Energy, Total Band
Nominal Zero (1% Distortion) on Tape
Level for 4 db Compression by Tape
Level for 2 db Compression by Tape
FREQUENCY IN CYCLES PER SECOND
Patricia (1½ min.)
Perez Prado Band.
Rhythm channel; prominent guitar, electric guitar and snare drums.
Recording from RCA: stereo, right hand channel only.
Master Tape.

FIG. 20.

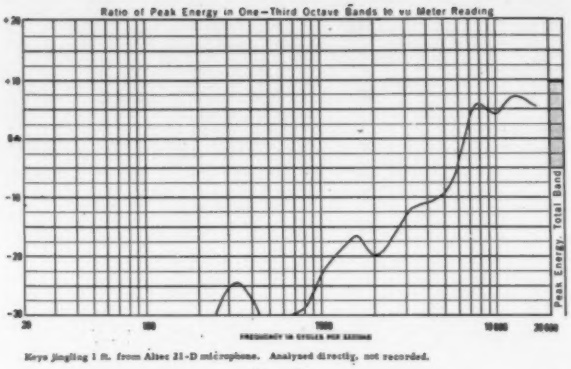
Ratio of Peak Energy in One-Third Octave Bands to vu Meter Reading
Peak Energy, Total Band
FREQUENCY IN CYCLES PER SECOND
Keys jingling 1 ft. from Altec 21-D microphone. Analyzed directly, not recorded.

FIG. 21.

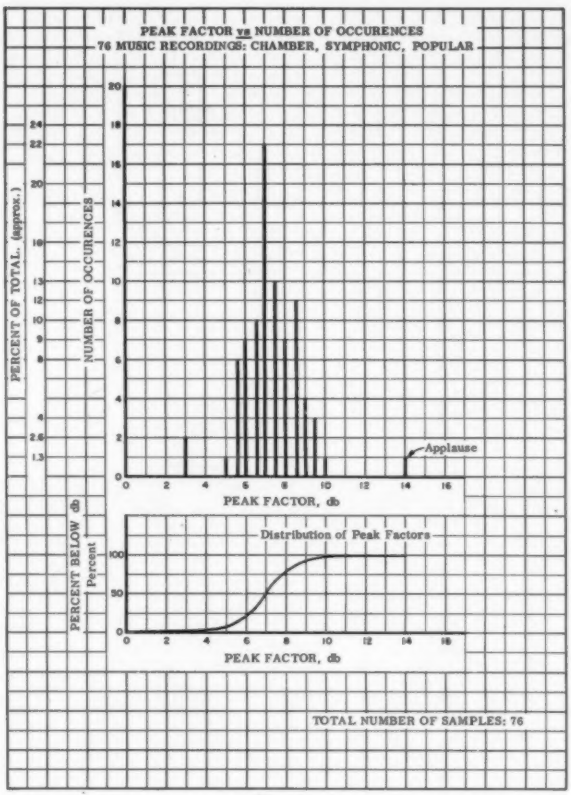

FIG. 22.

shows almost constant energy to 5 kc, falling sharply above. But the rhythm instruments channel (Fig. 20) has an 8 db peak at 3 kc, from a prominent electric guitar, and is down only 5 db at 8 kc.

Fig. 21, for interest, is keys jingling, one foot away from

an Altec 21-D microphone, analyzed directly (not recorded). It is easy to see why key jingling is hard to record and reproduce.

Although this paper does not discuss peak factors individually, Fig. 22 shows a summary of the peak factors of 76 music recordings. (The peak factor is the amount by which the true peak energy exceeds the VU meter reading.) The values of peak factor are seen to be from 3 to 10 db for music, with 7 db being the most common value. If in our equipment design we allow for peaks 10 db over the VU meter "zero" level, we will include all the music studied here. (This assumes one adjusts the level so that the VU meter usually swings to zero, with occasional peaks to plus one or two. Nothing is left for those who "pin the meter" frequently, as too many do.)

Fig. 23 shows several of the analyses drawn on one graph, to re-emphasize the variability in spectra from various recordings. *Realize especially that all of these give the same reading on a VU meter.*

Suppose we were to draw *all* of the analyses, in similar fashion, on one graph. Then we would get an area, as shown in Fig. 24. Again, all of the curves which would fall in this area give the same reading of zero on a VU meter. (The Sivian, Dunn, and White data comes near to the top of this area, also).

What do we conclude from all of this data? Mainly, that a usable "average" spectrum for music does not exist. If we have a record/reproduce system using a VU meter for its level indicator and if we are to allow for the maximum high- and low-frequency energy found in these studies, then the system must be able to handle all signals from at least 40 c to 16 kc at the VU "zero" level. In general, this means little use of pre-emphasis, if we are to be able to pass all frequencies at zero level.

Why does our present equipment work at all, having been designed on the assumption of much less high-frequency energy than this data shows? First, not all material has the maximum energy—many recordings will be in the middle to bottom of the area. Second, a fair amount of overload may exist without becoming objectionable. Third, it is common practice to listen to tapes, to see if overload does exist; if it does, the level is lowered. Fourth, we *don't* always get away with it! Fifteen-ips NAB equalized tape is almost always satisfactory, since it will pass all frequencies at zero level. But disc and 7½ ips tape often have noticeable high-frequency breakup. And 3¾ ips tape for music can be very

(*Continued on Page 80*)

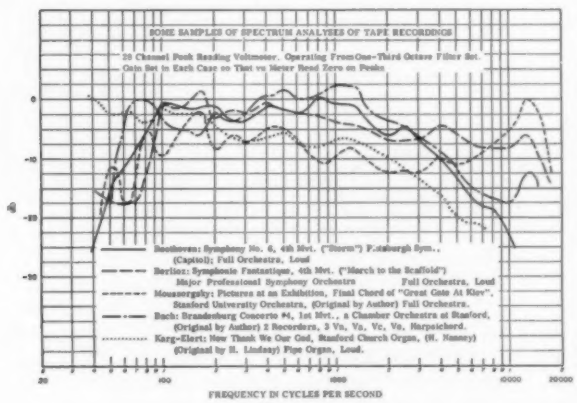
SOME SAMPLES OF SPECTRUM ANALYSES OF TAPE RECORDINGS
20 Channel Peak Reading Voltmeter, Operating From One-Third Octave Filter Set.
Gain Set in Each Case so That vu Meter Read Zero on Peaks
Beethoven: Symphony No. 6, 4th Mvt. ("Storm") Pittsburgh Sym., (Capitol); Full Orchestra, Loud
Berlioz: Symphonie Fantastique, 4th Mvt. ("March to the Scaffold") Major Professional Symphony Orchestra Full Orchestra, Loud
Mousorgsky: Pictures at an Exhibition, Final Chord of "Great Gate At Kiev". Stanford University Orchestra, (Original by Author) Full Orchestra.
Bach: Brandenburg Concerto #4, 1st Mvt., a Chamber Orchestra at Stanford, (Original by Author) 2 Recorders, 3 Vn, Va, Vc, Vo, Harpsichord.
Karg-Elert: Now Thank We Our God, Stanford Church Organ, (H. Nanney) (Original by H. Lindsay) Pipe Organ, Loud.
FREQUENCY IN CYCLES PER SECOND

FIG. 23.

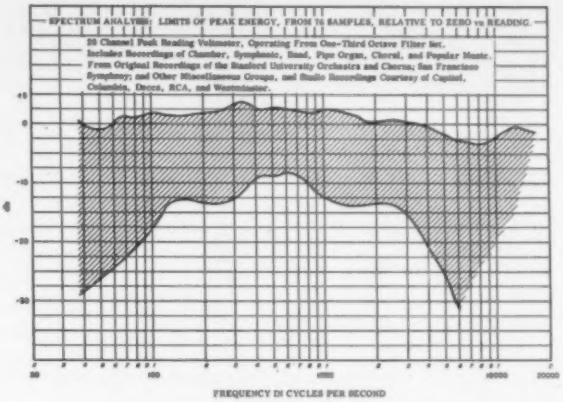
SPECTRUM ANALYSIS: LIMITS OF PEAK ENERGY, FROM 76 SAMPLES, RELATIVE TO ZERO vu READING.
20 Channel Peak Reading Voltmeter, Operating From One-Third Octave Filter Set.
Includes Recordings of Chamber, Symphonic, Band, Pipe Organ, Choral, and Popular Music.
From Original Recordings of the Stanford University Orchestra and Chorus; San Francisco Symphony; and Other Miscellaneous Groups, and Studio Recordings Courtesy of Capitol, Columbia, Decca, RCA, and Westminster.
FREQUENCY IN CYCLES PER SECOND

FIG. 24.

(*Continued on Page 80*)

THE AUTHOR

John G. McKnight, who was born in Seattle in 1931, studied at Stanford University and received his B.S. in Electrical Engineering there in 1952.

In 1953 he worked with Ampex Corporation on the development of cinemascope-stereophonic sound equipment. After several years of other work, he returned to Ampex, where he is now Senior Engineer in the Research Division. He spent the years 1953-1956 on the engineering staff of the Armed Forces Radio Service in New York. At the same time he worked as development engineer for the Gotham and the Narma Audio Development Companies. He has always been interested in the problems of magnetic recording, specifically as they concern music.

Mr. McKnight is a member of the Audio Engineering Society and an affiliate member of the IRE.

Psychoacoustics Applied to Stereophonic Reproduction Systems*

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CBS Laboratories, a Division of CBS, Inc., Stamford, Conn.

In an attempt to develop less cumbersome stereo speaker systems for home use, subjective listening tests were undertaken. A good two-speaker system was used as reference. Using crossover networks and delaying signals to speaker reduced the "precedence effect" and the listener's ability to assign particular frequencies to a particular speaker. A simple application of the principles to home use is illustrated.

THE WORK to be described was undertaken for the purpose of applying psychoacoustic findings to the development of stereophonic reproducing systems that would be more convenient and generally acceptable to the home user than the conventional one employing two full high-fidelity systems and speakers placed to each side.

Previous work by others had indicated that such a goal might well be achieved. The work of Snow dating back to the early stereophonic work of the Bell Telephone Laboratories indicated the influence of arrival time on stereophonic localization.[1] The "precedence effect" in sound localization as studied by Wallach, Newman, and Rosenzweig[2] (also known as the "Haas effect" after its investigator[3]) showed that sound tends to be localized in the loudspeaker from which it first arrives at the ear of the listener. These studies led to our belief that by delaying the signal to a speaker, its influence on the apparent source of sound might be minimized, even though it might handle the main part of the sound energy. Bogert[4] of the Bell Telephone Laboratories showed that delay could be utilized in stereophonic systems in a similar manner.

As another approach to our objectives, it was our belief that very low frequencies were relatively unimportant for stereophonic reproduction in small rooms. We therefore had reason to hope that one approach to simplifying the sound system would be to handle low frequencies with one speaker, while smaller speakers to each side could handle the high frequencies.

Our experiments were directed towards investigating the degree to which such effects could be utilized, and determining what the parameters should be as to amount of delay or cutoff frequency. Also, of course, the work was directed toward trials of stereophonic speaker systems less cumbersome than the traditional two large speakers.

The first experiment was arranged to determine to what extent low frequencies could be eliminated from the difference signal. To this end, a normal stereophonic system was set up, with a chain of matrixors converting into sum and difference signals and then back to left and right signals. This is shown in Fig. 1. An adjustable high-pass filter was inserted in the difference signal path to remove the low frequencies. When this was switched in or out, qualitative listening tests at first indicated that most listeners could detect no difference on stereo speech and music if frequencies below 300 cycles were eliminated from the difference signal. With more listener training and choice of program material, some shift was just perceptible. Even the most critical could notice no difference if frequencies below 200 cycles were eliminated. Since the apparent source of the very low-frequency sounds was disturbed in any case in usual rooms because of reflections and the standing wave pattern, it was felt that for all practical purposes the stereophonic component could be eliminated below 250 cycles.

A side note may be useful. Adding and subtracting circuits, using transformers or vacuum tubes, are well known

* Revised copy received January 5, 1959. Presented at the Tenth Annual Convention of the Audio Engineering Society, New York, September 29, 1958.

† President.

‡ Scientific Aide to the President.

1. W. B. Snow, "Effect of Arrival Time on Stereophonic Localization," *J. Acoust. Soc. Am.*, (vol. 26, November, 1954).

2. Hans Wallach, Edwin B. Newman, and Mark R. Rosenzweig, "The Precedence Effect in Sound Localization," *Am. J. Psychol.*, (July, 1949, pp. 3-5-336).

3. Helmut Haas, "Uber den Einfluss Eines Einfachechos auf die Horsamkeit von Sprache," *Acustica*, (Vol. 1, 1951, pp. 2-58).

4. B. P. Bogert, "Stereophonic Sound Reproduction Enhancement Utilizing The Haas Effect," *J. Soc. Motion Picture Engrs.*, (vol. 64, June 1955, p. 308).

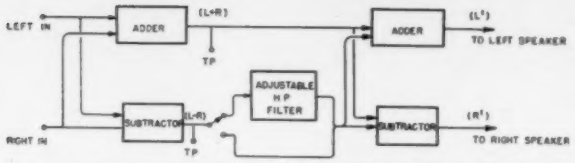

FIG. 1. Experimental Set-Up for Removing LF from Difference Signal.

in the art, but the resistive bridge matrixor may be less familiar. Fig. 2 shows an adding and subtracting circuit of this type. This has several advantages. Complete isolation between the two signal sources, shown as R and L, is obtainable if everything is balanced, 50 db being attainable in practice with precision components. Balance is no longer dependent on transformer perfection. However, the output circuit for $\frac{1}{2}(R + L)$ or $\frac{1}{2}(R - L)$ must truly represent an accurate resistive bridge arm for balance, and since one signal source cannot be grounded, an isolation transformer may be required, but good quality is more readily achieved for this than for a well balanced multi-winding transformer.

The second experiment was to determine the practicality of using one large speaker to handle the low frequencies,

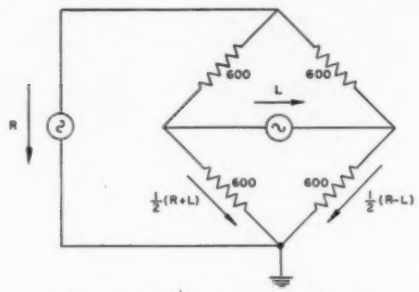

FIG. 2. Resistive Bridge Matrixor.

while two small speakers or satellite speakers take care of the frequencies above 250 cycles and contribute the stereophonic reproduction. Fig. 3 shows the arrangement. Crossover networks were used to divide the frequency range. The woofer consisted of two in-phase speakers in one box, independently fed. Single-speaker woofers were also tried, the low-frequency crossover outputs being connected in parallel. Performance was determined qualitatively by comparison with a two-speaker stereo system, an effort being made to match the tone quality. About two milliseconds of delay is introduced by the low-frequency crossover network. Results were very encouraging, any difference between the two systems being attributable more to tonal quality differences than to stereophonic behavior.

The third experiment, shown in Fig. 4, used similar speaker arrangements, but inserted crossover networks at a low level, adding left and right signals so that the L.F. sum signal was connected to the woofer, while the H.F. right and left signals went to the side speakers. Independent amplifiers were used ahead of each speaker. In this way, smaller and less expensive crossover networks could be used, possibly RC networks, and the satellite speakers could use

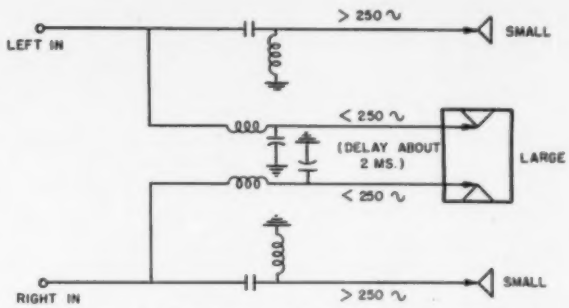

FIG. 3. Experimental System with 2 Amplifiers, 3 Speakers.

smaller amplifiers, only one amplifier being needed with a large and high quality output transformer to provide high-level low-frequency output to the woofer.

The mixed low-frequency path was fitted with both a low-pass filter and an adjustable delay line, in order to evaluate the extent to which delay to the woofer contributes to the satellite speakers controlling the stereophonic effectiveness. Qualitative listening tests were again employed for comparing the system to a good two-speaker reference system. The tone quality was quite well matched, using individual frequency-response adjustments for the separate amplifiers. Half-octave hands of white noise were used in tone matching, instead of warbled oscillator tones as used previously, except for the very high frequencies. The system also allowed for insertion of low-pass filters of differing cut-off slopes, in order to see how complex a low-pass filter was required.

It was found that no difference could be observed between delays of about 7 to 20 milliseconds, and that delay of 2 milliseconds normally provided by the cutoff network was

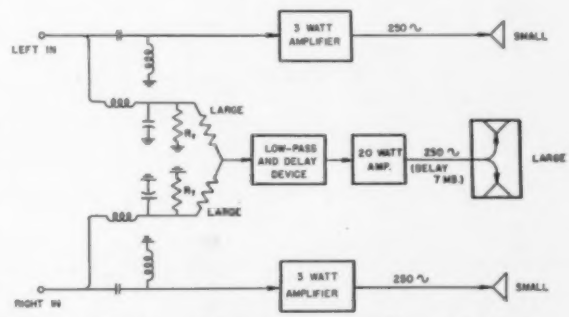

FIG. 4. Experimental System with 3 Amplifiers, 3 Speakers.

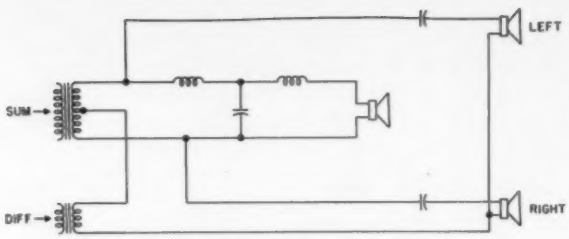

FIG. 5. Application to Home Units.

very little inferior in stereophonic effect unless the listener was located much closer to a side speaker than to the woofer, so that the air path length difference counteracted the delay. We had hoped to try zero and negative delays by using independent heads on a magnetic tape system, adjustable in relative position to provide adjustable delay. In this way, the question as to which contributes most to the stereophonic effectiveness of the system, the delay or the low-pass filter action, could have been resolved. This was not carried out, and further work could be done in this direction.

As to the required rate of cutoff slope in the low pass filter, it was found that 36 db/octave was no more effective than 24 db/octave, and little if any difference was observed between 12 and 24 db/octave except for especially critical tests such as seeing if any pure frequency above 250 cycles could give an appreciable output from the woofer compared to the small speakers. 12 db/octave would usually be satisfactory.

The main result of the tests was to show that a woofer with two small satellite speakers could be made to give stereophonic performance essentially equal to that of a system using two large separate speakers and amplifiers. The woofer could be moved laterally quite far to one side before stereophonic effectiveness was degraded, indicating that an alternative to the woofer and satellites could be a full-range system on one side and one satellite on the other. It was interesting to see visitors walk up to a small speaker, convinced that the low-frequency sounds also were coming from the little satellite, until they were only a foot or so away.

The next step was the application of these principles to a home system. Fig. 5 shows how simply this works out in practice. Sum and difference inputs are shown. With a different output transformer arrangement, left and right inputs can be used instead. Transformer addition and subtraction yields left signal and right signal for the satellite speakers, connected through capacitors acting as high frequency crossover "networks." Right plus left, *i.e.*, the sum signal, is applied to the woofer through a low-pass filter cutting off at about 250 cycles, and having associated delay. In a companion paper[5] Bauer has shown that improved stereophonic area coverage can be obtained by substituting "Isophonic" satellite speakers in place of the usual small speakers covering 250 c and above.

5. B. B. Bauer, "Broadening the Area of Stereophonic Perception" presented before the Audio Engineering Society, September 29, 1958.

THE AUTHORS

PETER C. GOLDMARK

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Dr. Peter C. Goldmark was born in Budapest in 1906 and studied at the Universities of Berlin and Vienna, receiving his Ph.D. in physics from the latter. From 1931-33 he was Engineer in charge of Television Development for Pye Radio Limited, England. In 1933 Dr. Goldmark left England for the United States and in 1936 joined Columbia Broadcasting System as Chief Television Engineer, later becoming Director of the Research and Development Division. The long-playing record was developed by Dr. Goldmark and his associates in the CBS Laboratories. Later the development of the first high fidelity compact phonograph, the Columbia 360, followed.

In 1954 Dr. Goldmark became President of CBS Laboratories and Vice President of CBS Inc. He is a Fellow of the American Institute of Electrical Engineers, the Institute of Radio Engineers, the Society of Motion Picture and Television Engineers, and the British Television Society. In 1945 Dr. Goldmark was awarded a medal by the Television Broadcasters Association for his color television pioneering work, and in 1946 the Institute of Radio Engineers awarded him the Morris Liebman Memorial Prize for electronic research.

Mr. Hollywood received B.S. and M.S. degrees in Electrical Engineering from M.I.T. in 1931 and 1932. From 1933 to 1936 he worked in cathode ray tube development and associated circuitry. In 1936 he joined CBS, working primarily in color television development. During the war he served in England with the Radio Research Laboratory of Harvard University. In 1946 he joined the Naval Research Laboratory as consultant, and in 1947 the Airborne Instruments Laboratory.

In 1949 he rejcined CBS Laboratories, where he is presently Scientific Aide to the President. He has a number of patents in various fields to his credit, has written articles on electronic subjects, and is the author of a chapter of the *Television Engineering Handbook* edited by Donald Fink. He is a member of the Audio Engineering Society, the Acoustical Society of America and Senior member of the Institute of Radio Engineers.

Designing a Multi-Purpose Stereo Pre-Amplifier*

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In between the required sensitivity and the required output voltage the stereo pre-amplifier should be designed with a good tone and volume control system. Logical design leads to a pre-amplifier with distortion as low as 0.03% for all frequencies. Included in the design is a simple but effective rumble filter.

SPECIFICATION

WE SHALL assume that we have to design a pre-amplifier according to the following specifications:

OUTPUT

Voltage (for full drive of power amplifier): 0.6 V.

Impedance: suitable for short cable connections to power amplifier.

INPUT

Tape: Full output for 60 mV equalized input } High impedance (1 MOhm)

Tuner: Full output for 350 mV input }

Pick-up: a) Dynamic with full equalization according to RIAA standard full output for 5 mV input at 1000 c/s.

No increase in distortion for input voltages up to 200 mV at 1000 c/s.

Input impedance 100 k Ohm at 1000 c/s.

b) Crystal with full equalization according to RIAA standard full output for 150 mV at 1000 c/s.

No increase in distortion for input voltages up to 1 V.

Input impedance 100 k Ohm.

Switching between *a* and *b* at rear of amplifier.

All inputs for both monaural and stereo.

Rumble filter with sharp cut-off at 25 c/s.

CONTROLS.

Volume control with physiological bass compensation for levels lower than – 30 db (re top).

Treble control ± 10 db at 10 kc/s.

Bass control ± 10 db at 50 c/s.

Left-right adjustment 0 to – 14 db.

1. GENERAL CONSIDERATION

The design in general is based on the consideration that there are two categories of input signals: those from a pick-up, being of low level and requiring equalization; and those from a tuner or a tape-deck (the latter including the pre-amplifier) giving a "flat" signal of a much higher level. Therefore, as is customary for pick-up signals, additional amplification is required, which according to the specification should be based on an input signal of 5 mV at 1000 c/s. In reality, however, the available amplification must be considerably more because of the RIAA recording characteristic which asks for 18 db more sensitivity at 50 c/s. Therefore the actual amplification for the pickup signal should be based on an input signal of 0.6 mV.

The signal from the tape recorder, however, is of the order of 60 mV. If we bring the pickup signal to an (equalized) level of 60 mV we can handle both types of signals from there on in the same way. For the equalizing and amplification of the pick up signal one pentode in the appropriate circuitry is sufficient.

Thus we have divided our pre-amplifier into two parts, the first one containing one pentode for amplification and equalization, whilst the other part comprises all controls and further amplification necessary to obtain the required level of 0.6 V.

The pentode chosen in the first part is an EF86. This tube, which is especially designed as a low noise low microphony af amplifier tube, is extremely suitable for amplifying the very small input signals because the extra noise and hum which it introduces is negligibly small.

For the second part one other tube, a double triode, is necessary and sufficient for each channel. This statement makes allowances for the tone control system chosen.

The tone control system can be either passive or active, the frequency discriminating elements of the latter usually being included in a feedback loop. A disadvantage of the feedback tone control is that the distortion figure depends on the frequency and on the position of the controls. An

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† Manager, Reproduction Advisory Group.

advantage is that slightly higher amplification factors can be obtained.

Our final decision was to use a passive tone control. We reached this conclusion by the following reasoning. We have a signal of 60 mV at our disposal which ultimately should lead to an output signal of 600 mV. But, when calculating the necessary amplification, we have to allow for an attenuation for the plus position of the tone controls (a factor 5). The required amplification factor is therefore $\frac{600}{60} \times 5 = 50$.

Now we could have used one pentode, with the tone control elements in a feedback circuit. This method, however, is not advisable, because feedback between anode and grid (as would have been the only possible procedure) is rather awkward because it leads to low input impedance and gives little possibility for negative feedback in the maximum position of the tone controls. The other possibility is to use two triodes, *e.g.*, a double triode. This method renders so much amplification that the use of a passive tone control is indicated, a fair amount of frequency-independent feedback in the double triode being available but leading to extremely low distortion figures. Therefore the basic system of the pre-amplifier is now determined to be as follows: per channel a pre-amplifier/equalizer EF86 and one double triode (ECC83) with a fair amount of feedback. In between those tubes are the balance and the volume controls, and at the end of each amplifier are the passive tone control systems.

2. DETAILS

a. The equalizer-amplifier

Because the voltages generated by a dynamic pickup are in the mV-range, a low-hum low-noise tube is required. We found the EF86 extremely suitable for this purpose. The RIAA correction can easily be obtained by means of a simple negative feedback network between anode and control grid. The amplification obtained is sufficient to allow also for some negative feedback in the very low frequency range. The output voltage of this tube is so small (in the extreme case 2½ V) that the distortion is negligible.

The amplification is such that a 5 mV input signal of 1000 c/s gives an anode voltage of 60 mV, sufficient for the ECC83 to give the required 0.6 V necessary for fully driving the output power amplifier. On the other hand even input signals of 40 times this value can be handled without overloading the first tube. This feature is obtained by choosing proper setting of the EF86. The necessity of a large overloading factor is very often overlooked.

The input impedance for high frequencies is low, so that a fairly long screened connection between pickup and input terminals may be used.

b. The left-right control

The left-right control is effected by a special double potentiometer. At the center position both channels receive their maximum input signal. If the control is turned in one direction, one signal is attenuated whilst the other remains constant. Turning the control the other way around causes the opposite to happen. The control characteristic is shown in Fig. 3.

c. The volume control

This is a one megohm potentiometer with a tap at 200 kOhm. The 200 kOhm section is shunted by a RC combination with the 3 db point at 400 c/s giving bass compensation for lower levels as specified.

d. The double triode section

As double triode a ECC83 was chosen. We have already seen that the required amplification of this tube for 60 mV input is 50 times. This value allows for considerable feedback. Under these conditions and with appropriate grid bias and anode load resistors, the output of the ECC83 can be as high as 10 V before distortion starts to increase noticeably.

Because the tone controls give an attenuation of a factor 5 for the center frequency of 1000 c/s the available output voltage is 2 V. The distortion at an output of 1 Volt is 0.03% for all frequencies between 30 and 20,000 c/s, independent of the position of the tone controls.

e. The rumble filter. (For calculation see Appendix.)

The rumble filter which is incorporated in the feedback and coupling system of the double triode is very simple and very effective. Its functioning is based on the following consideration: a 2-stage resistor-capacitance coupled amplifier can give a roll-off towards the lower frequencies with an ultimate slope of 12 db/octave. This ultimate slope, however, is reached very gradually, because the system has a Q of 0.5. By applying negative feedback the Q can be increased. With a higher Q the ultimate slope is reached much more suddenly. The most useful value is $Q = 1$, because the small rise occurring then just before the roll-off can be compensated by means of another RC member, increasing the slope at the same time to 18 db/octave. A relation can be found between the roll-off frequency, the feedback factor, the Q of the circuit and the RC products. For a given frequency a certain feedback factor and assuming $Q = 1$, we find in general two different values for the time constants of each of the coupling elements (see appendix). It is desirable to apply the smallest time constant to the first stage because this has of course the lowest anode voltage, allowing for the unavoidable rise at the lower frequencies at the anode side of the coupling condenser. If the smallest coupling condenser had been placed in the output of the second half of the ECC83 then, to obtain the required

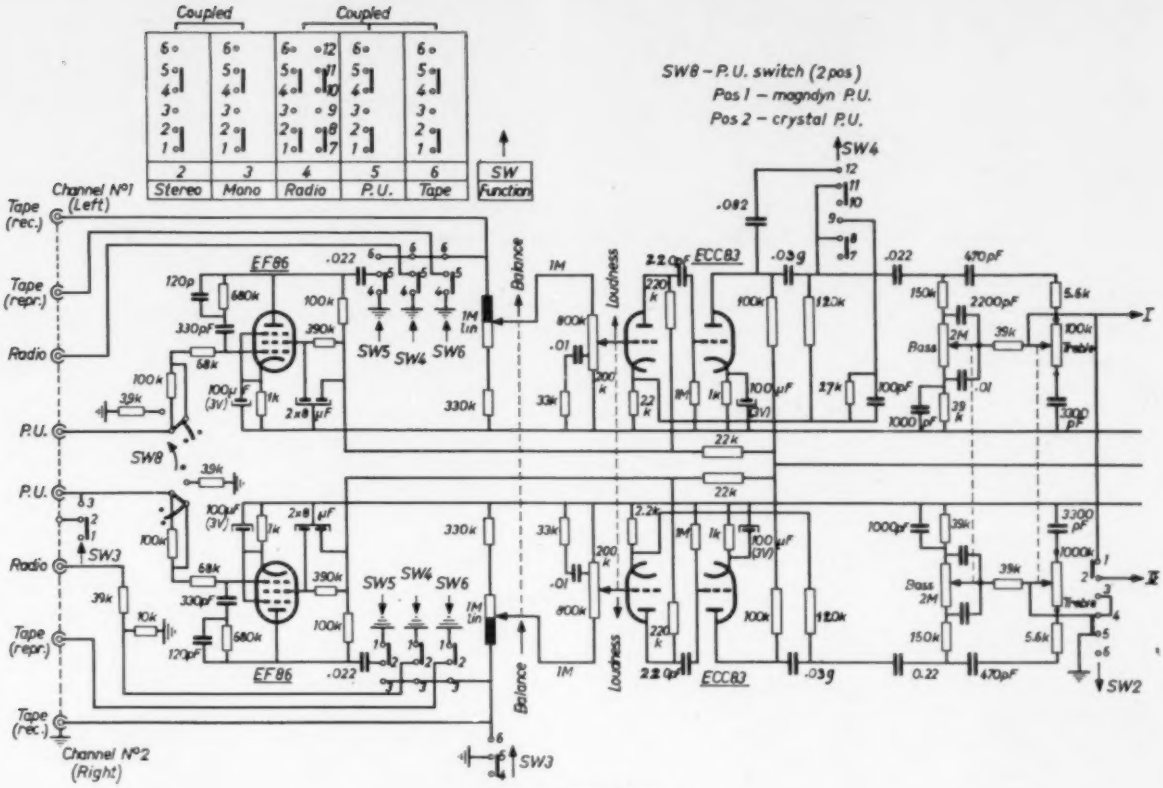

FIG. 1. Circuit diagram.

flat response slightly above the cutoff frequency, the anode voltage would have been much higher, resulting in excessive distortion at the low frequency end.

f. The tone control system

The tone control system is one of the well-known circuits. The value of the components is chosen to give a good compromise between the maximum range of the controls and the gradual changing of the characteristic as a function of the angle of rotation of the knobs. The way in which the components are arranged gives a fairly low output impedance for the higher frequencies, allowing the use of reasonably long connecting leads.

g. Decreasing the sensitivity for tuner input

Input signals from the tuner can be expected to be four times higher than those of other sources. Therefore a special input for higher voltage is provided, the decreased sensitivity being obtained by increased negative feedback in the ECC83. In this way the signal-to-noise ratio is kept as favorable as possible.

h. Crystal pickup input

A crystal pickup is an amplitude pickup by nature, but it can be given velocity characteristics (and therefore made fit for use with standard equalizer circuits) by connecting an appropriate resistor across its terminals.

This system, together with a potential divider for obtaining the right sensitivity, has been chosen in this amplifier.

j. The input selecting switch

The main input switching is done by push buttons on the front panel. There is a choice from three input sources: pickup, tape, and tuner. Each of these inputs can be used either stereophonically or monaurally. In the latter case the two sets of output terminals receive the same signal, so that the two power amplifiers connected to these terminals are driven in parallel.

The switch for changing over from crystal to dynamic pickup is at the rear.

k. The complete circuit diagram clearly shows the electrical layout of the circuit (Fig. 1).

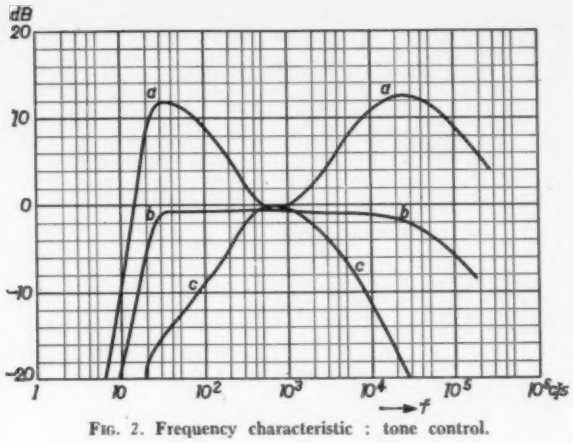

FIG. 2. Frequency characteristic : tone control.

3. MEASURED RESULTS

a) Frequency characteristic: tone controls (Fig. 2).
b) Left right control (Fig. 3).
c) Equalization (Fig. 4).
d) Distortion at 1 V output (independent position of tone control) 0.03%.
e) Hum and noise.
   1) Selector switch in position pickup under operating conditions (left-right in center position, tone controls flat, volume control adjusted for maximum output for 20 mV input at 1000 c/s) unweighted hum and noise –60 db.
   2) Selector switch in other positions.
   Left-right control in center position. Unweighted hum and noise > –60 db.
f) Crosstalk.
   At 1000 c/s > 50 db.

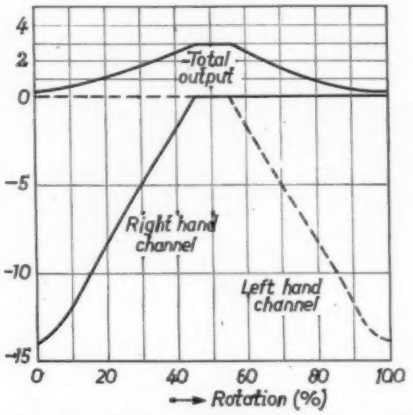

FIG. 3. Left-right control. Attenuation per channel as a function of rotation of the knob.

APPENDIX

The designing of the rumble filter

We first determine the gain and frequency response of the stage without overall feedback.

The connection between R_0 and R_2 (Fig. 5) is loosened and R_2 is connected to ground. The equivalent circuit is seen in Fig. 6, in which the two triodes are replaced by two current sources and internal resistances.

The amplification is:

$$A=\frac{V_2}{E}=\frac{-g'_{m1}\cdot R_{p1}}{\frac{1+1}{j\omega T_1}}\cdot\frac{g_{m2}\cdot R_{p2}}{\frac{1+1}{j\omega T_2}}=\frac{A_{1.2}}{\left(\frac{1+1}{j\omega T_1}\right)\left(\frac{1+1}{j\omega T_2}\right)}. \quad (1)$$

$A_{1.2}=g'_{m1}\cdot R_{p1}g_{m2}R_{p2}$ amplification at high frequency ($\omega\rightarrow\infty$) (2)

$T_1=C_1\left(R_1+\frac{R'_{i1}\cdot R_{a1}}{R'_{i1}+R_{a1}}\right)$ time constant of the first triode (3)

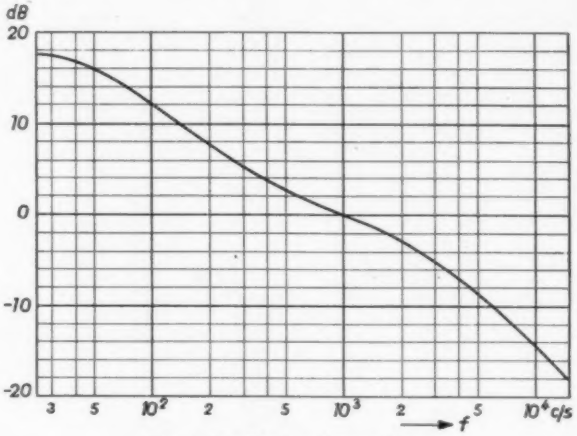

FIG. 4. Equalization characteristic.

$T_2=C_2\left(R_2+\frac{R_{i2}\cdot R_{a2}}{R_{i2}+R_{a2}}\right)$ time constant of the second triode (4)

$R'_{i1}=R_{i1}+\mu_1 R_0$ internal resistance of the first triode as a result of the current feedback ($\mu_1 >> 1$).

$g'_{m1}=\frac{g_{m1}}{1+g_{m1}R_0}$ dynamic mutual conductance of the first triode as a result of the current feedback.

R_{p1} parallel resistance of $R'_{i1}/R_{a1}/R_1$.

R_{p2} parallel resistance of $R_{i2}/R_{a2}/R_2$.

Investigating the frequency response, making $T_1=T_2=T$, from equation (1) we see, there are two CR sections in cascade, with a 3 db roll-off point at the same frequency.

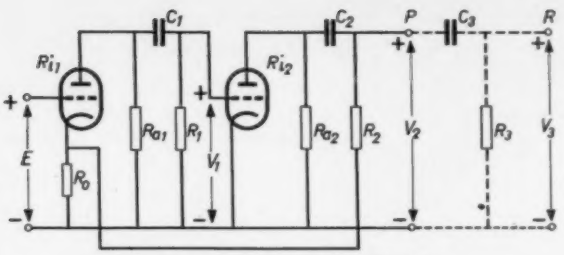

FIG. 5. Basic circuit of the rumble filter.

at high frequency ($\omega \to \infty$) : $A = A_{1.2}$ constant.

at low frequency ($\omega \to 0$) : $A = -A_{1.2}(\omega_T)^2$ falling down with 12 db/oct.

From Fig. 7, Curve a, we see that the slope between these ultimate frequency ranges is smaller than 12 db/octave in a rather large frequency range. We may improve this by using overall feedback, which will be positive in a certain frequency, resulting in a peak in the frequency response.

$$A' = \frac{A}{1 + A\beta}. \tag{5}$$

with $\beta = \dfrac{R_0}{R_2 + R_0}$ feedback ratio.

Substituting equation (1) for A, and by introducing a new frequency parameter $\gamma = \dfrac{\omega}{\omega_o}$ we find:

$$A' = \frac{A_{1.2}}{N} \cdot \frac{jQ\gamma}{1 + jQ\,(\gamma - 1/\gamma)} \tag{6}$$

with $\gamma = \omega/\omega_o$ frequency parameter.

$$\omega_o = \frac{1}{\sqrt{NT_1T_2}} \text{ cutoff frequency.} \tag{7}$$

$$N = 1 + A_{1.2}\beta \text{ Nyquist factor.} \tag{8}$$

$$Q = \frac{\sqrt{NT_1T_2}}{T_1 + T_2} \text{ Quality.} \tag{9}$$

The frequency response

At high frequency ($\gamma \to \infty$) : $A' = \dfrac{A_{1.2}}{N}$ constant. (10)

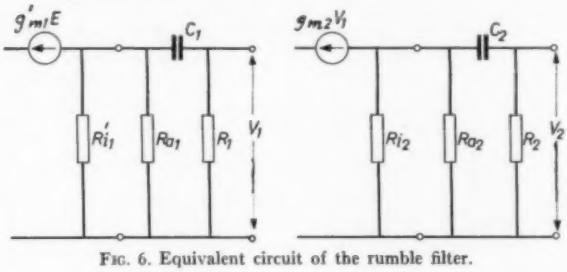

FIG. 6. Equivalent circuit of the rumble filter.

At low frequency ($\gamma \to 0$): $A' = -\dfrac{A_{1.2}}{N}\gamma^2$ falling down with 12 db/oct.

At $\omega = \omega_o$ ($\gamma = 1$): $A' = jQ\dfrac{A_{1.2}}{N}$. (11)

There is a peak in the frequency response if $Q \geq 0.7$. The larger Q, the larger the peak. Making the first derivative in respect to γ of equation (6) zero, we find a maximum at $\gamma = 1$, for high Q's. For Q's of about 1, the frequency is somewhat higher. If we choose a $Q = 1$, there will be a small peak in the frequency response and then a sudden fall with 12 db/octave (Fig. 7, Curve b).[1] Combining this filter with a single CR section, with a roll-off point (3 db) at $\omega = \omega_o$, the peak will be compensated and a flat response curve with a sharp roll-off at $\omega = \omega_o$ with 18 db/octave, will be the result (Fig. 7, Curve c).

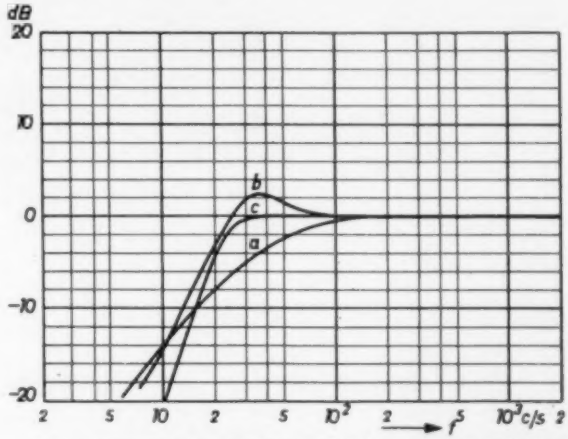

FIG. 7. Frequency characteristics of 2 RC members. a. Roll-off at -3db point is much less than 12 db/octave. b. When applied in appropriately designed feedback circuitry (Q = 1) a small peak occurs followed almost immediately by a 12 db/octave slope. c. Adding another RC member removes the small peak and gives ultimately an 18 db/octave curve.

$$\omega_o T_3 = 1 \quad (T_3 = C_3R_3). \tag{12}$$

Choice of the elements

$R_{i1} \sim R_{i2} \sim 60.10^3$ ohm : $\mu_1 \sim \mu_2 =$ 100 : $g_{m1} \sim g_{m2} \sim 1{,}6.10^{-3}$ A/V.

$R_{a1} = 220.10^3$ ohm : $R_1 = 10^6$ ohm : $R_{a2} =$ 10^5 ohm : $R_0 = 2{,}2.10^{-3}$ ohm.

$g'_{m1} = \dfrac{g_{m1}}{1 + g_{m1}R_0} = 0{,}35.10^{-3}$ A/V : $R'_{i2} =$ $R_{i1} + \mu_1 R_0 = 280.10^3$ ohm.

[1] The formula from which the frequency is found at which a peak occurs is: $\eta = \dfrac{Q\sqrt{2}}{\sqrt{2Q^2 - 1}}$. From this it will be clear that no peak can occur if $Q < \frac{1}{2}\sqrt{2}$.

From (9): $A' = \frac{A_{1.2}}{1 + A_{1.2}\beta} \sim \frac{1}{\beta}$ if $A_{1.2}\beta >> 1$.

The gain of the tube has to be 50 in this case.

Therefore, $\beta = \frac{R_0}{R_0 + R_2} = \frac{1}{50}$ and $R_2 = 112.10^3$ ohm.

We used: $R_2 = 120.10^3$ ohm.

With the given values and equation (2), we may solve $A_{1.2} \sim 1170$ and from (8) the Nyquist factor $N = 26.4$.

From the equations (7) and (9), we may solve T_1 and T_2 if the cutoff frequency and N are known, making $Q = 1$.

$$T_1 = \frac{1}{4\pi f_o}\left[1 \pm \sqrt{1 - \frac{4}{N}}\right] \qquad (7a)$$

$$T_2 = \frac{1}{4\pi f_o}\left[1 \mp \sqrt{1 - \frac{4}{N}}\right] \qquad (9a)^2$$

Substituting $f_o = 25$ c/s and $N = 26.4$ we find:

$T_1 = 0{,}25.10^{-3}$ sec. (or $6{,}1.10^{-3}$ sec.)

$T_2 = 6{,}1.10^{-3}$ sec. (or $0{,}25.10^{-3}$ sec.)

From equations (3) and (4), C_1 and C_2 are solved:

$C_1 = 222$ *pF*. We used: $C_1 = 220$ *pF*

$C_2 = 38{,}5$ *nF*. We used: $C_2 = 39$ *nF*

R_3 consists of the bass potmeter of 2 MOhm and the resistors of 150 and 39 KOhm in series. Substituting this value for R and $\omega_o = 2\pi 25$ in (12) we may solve C_3:

$C_3 = 29$ *nF*. We used: $C_3 = 22$ *nF*, being the nearest practical value.

NOTE: It is desirable to use the smallest T in the first triode, for when using it in the second triode, the voltage on the anode of this triode will increase too much, as the output voltage of the second stage as to be constant up to 25 c/s causing distortion. The voltage on the anode of the first triode is smaller.

[2] Because a time constant should be a real number the factor N should comply with the condition $N \geq 4$.

THE AUTHORS

J. RODRIGUES DE MIRANDA

H. VAN DEN KERCKHOFF

J. Rodrigues de Miranda, who was born in 1905, received his schooling in Holland. In 1927 he graduated from the technical University of Delft as an electrotechnical engineer. He spent the next eleven years managing the technical department of a wholesale importing company. He joined N. V. Philips' in 1938, and he has remained there doing development work in acoustics. He is now the Senior Engineer and Manager of the Philips' Reproduction Advisory Group. His memberships in numerous professional societies include the AES and IRE in the United States.

H. van den Kerckhoff, born in 1926 in Indonesia, joined Philips' and Mr. Miranda in 1956, the year he graduated from the Technical University at Delft. In his laboratory investigations on acoustics, he has concentrated on the development of amplifiers and preamplifiers.

(*Continued from Page 71*)

bad, if a great deal of high-frequency energy is present, or very good if high-frequency content is controlled.

CONCLUSION

Does this data indicate the need for any action on the part of magnetic recording equipment design engineers? If our present recording equipment had signal-to-noise ratio to spare, we could simply allow for the worst case at all times. Since we do not have this spare signal-to-noise ratio, we can pursue the alternate course mentioned at the beginning of this paper: namely, design a post-emphasis for minimum audible noise; then use a level indicator which will insure that the system will not be overloaded. Such a device *might* be the equalized, peak-reading volume indicator. This, however, has the disadvantage that the meter does not indicate level balance. But the VU meter does not indicate overload —and we must either decide which is more important or use two meters, one for balance, one for overload. (What we propose is not new, but has not been widely applied as yet.) This would in any case require additional complication and work on the part of the manufacturer and the operators, but we feel that this might lead to a method for increasing the signal-to-noise ratio without necessarily increasing the distortion of the recording. Considerable evaluation work is needed to see if this method of metering will fulfill its promise.

Time Errors in Magnetic Tape Recording*†

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A theoretical treatment of time errors encountered in magnetic tape recording is developed. Effects of random noise are considered, as well as motional effects (flutter). Experimental results, obtained using an Ampex Model 3078 two-channel tape recorder and an Axis-Crossing Interval Meter, are presented. The use of autocorrelation methods to predict and measure flutter characteristics is discussed in detail.

INTRODUCTION

MAGNETIC TAPE recording is used extensively in many fields for the recording and reproduction of experimental data. For many applications, the small amounts of amplitude fluctuation and time error encountered in conventional data recorders can be tolerated. In applications requiring an unusually high degree of accuracy in recording and reproducing amplitude and phase information, however, severe restrictions are placed on the types of data storage and playback facilities that can be utilized.

During the last few years, a great deal has been accomplished toward reducing spurious amplitude fluctuations, chiefly through improvements in the quality of magnetic tapes. Some reduction of time errors has also been achieved, notably through the design of special data recorders such as the Ampex Model 500. Digital recording techniques may, of course, be employed to achieve nearly perfect reproduction; however, the emphasis here is on time errors experienced in conventional "analog" recording. It is felt that such studies might prove useful to those considering technical application of tape recorders. In addition, the theoretical approach employed may be directly applicable to problems of a similar nature in other fields.

TIME ERROR DETECTION

Nature of Time Errors

Time errors caused by flutter (irregular tape motion) and by random noise are to be considered in this paper. The random nature of these errors has been pointed out by a number of authors.[1] The signal at the output of an appropriate time error detector is a randomly fluctuating signal governed by a certain set of statistics. The statistical properties of fluctuating signals have been studied extensively during recent years,[2] and it is from this viewpoint that our theory is to be developed.

Axis-Crossing Interval Meter

The instrument employed to detect time errors in the experiments reported was the axis-crossing interval meter (ACIM).[3] The ACIM is a type of FM discriminator which produces a voltage proportional to the time interval between successive positive-going axis-crossings of an alternating input signal. The block diagram and principle of operation for the ACIM are presented in Figures 1 and 2. For a pure tone at its input, the ACIM produces a constant DC voltage. If the input is a more general alternating signal, the ACIM produces a series of flat-topped pulses: the amplitude of a given pulse is proportional to the duration of the preceding "cycle" of the input signal; the duration of a given pulse is equal to the duration of the concurrent cycle of the input signal. In practice the amplitude of a given pulse is proportional to the difference between the duration of the preceding cycle of the input signal and a preset reference time interval.

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* Delivered before the Ninth Annual Convention of the Audio Engineering Society, New York, October 8, 1957.

† The research reported in this paper was conducted at the Acoustics Laboratory, Massachusetts Institute of Technology, under a contract with the Bureau of Ships. The results were presented in the thesis: "Flutter in Magnetic Tape Recording," R. H. Prager, M.I.T. Department of Electrical Engineering, 1955.

‡ Electronic Scientist.

[1] For example, see:
H. N. Morris, "Measuring Magnetic Tape Recorder Flutter," *Electronics*, **26**, 230 (March 1953);
John T. Mullin, "Measurement of Flutter and Wow in Magnetic-Tape Instrumentation Recorders," *J. Audio Eng. Soc.*, **3**, 151 (1955).

[2] For example, see: Y. W. Lee, "Application of Statistical Methods to Communication Problems," M.I.T. Research Lab. of Electronics, *Report No. 181*, (September 1950).

[3] See R. H. Prager, *op. cit.*, for circuit diagram. See also: M.I.T. Acoustics Laboratory, *Quarterly Progress Report*, p. 32, (July-September 1951).

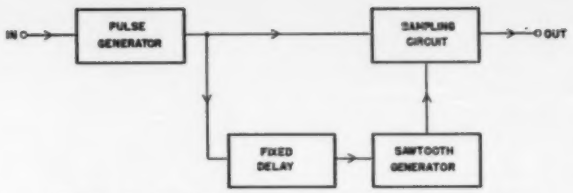

FIG. 1. Block diagram of Axis-Crossing Interval Meter.

For these measurements, a pure tone was recorded on the tape and the ACIM was adjusted to give zero DC output for the frequency recorded. The output voltage of the ACIM varied in accordance with variations in the period of the recorded signal upon playback. Since the time errors involved are small, the input to the ACIM was very nearly a pure tone; in such cases, variations in duration of the ACIM output pulses may be neglected. It is noted that the reported experiments were concerned with flutter frequencies higher than 1 kc. This restriction to relatively high-frequency time-error effects applies to the present paper as well.

THEORETICAL TREATMENT

Waveform Analysis

The signals to be considered are finite sections of tape playback and may be assumed to have finite energy content. That is, if $u(t)$ denotes a waveform (signal amplitude as a function of time), we may write

$$\int_{-\infty}^{\infty} u^2(t)\,dt < \infty.$$

The frequency amplitude (density) spectrum $U(f)$ of the waveform is given by the Fourier transform equation

$$U(f) = \int_{-\infty}^{\infty} u(t) \exp(-2\pi i f t)\,dt \qquad (1a)$$

and the waveform is regained through the inverse transform

$$u(t) = \int_{-\infty}^{\infty} U(f) \exp(2\pi i f t)\,df. \qquad (1b)$$

The waveform $u(t)$ and its associated spectrum $U(f)$ are referred to as a Fourier transform pair. For the purposes of this paper, $u(t)$ is always real; thus, its complex conjugate $u^*(t)$ is equal to $u(t)$. The spectrum $U(f)$ may be complex; it follows from equation (1a) that its complex conjugate $U^*(f)$ is equal to $U(-f)$.

The energy (density) spectrum of the waveform is defined as

$$C(f) = |U(f)|^2 = U(f)\cdot U^*(f) = U(f)\cdot U(-f). \qquad (2)$$

It may be seen from equation (2) that $C(f)$ is always real, positive, and an even function of frequency.

The (finite) autocorrelation function of the waveform is defined by

$$c(\tau) = \int_{-\infty}^{\infty} u(t)\,u(t-\tau)\,dt \qquad (3)$$

where τ is time delay. The autocorrelation function is a measure of the degree to which the fluctuations in two samples of the waveform $u(t)$ remain consistent as the samples are taken further and further apart in time. $c(\tau)$ is real and an even function of τ.

Some additional basic properties of $c(\tau)$ are:

a. The value at $\tau = 0$ is the total energy of the waveform. No other value exceeds $c(0)$.
b. The autocorrelation function of a periodic waveform is periodic in τ. It is apparent that if $u(t)$ is a pure tone, $c(\tau)$ has the same frequency in τ as $u(t)$ has in t.
c. The autocorrelation function of a sum of independent components is the sum of the autocorrelation functions of the individual components.
d. Autocorrelation of a random process with hidden periodic components yields distinguishable periodic and aperiodic functions of τ.

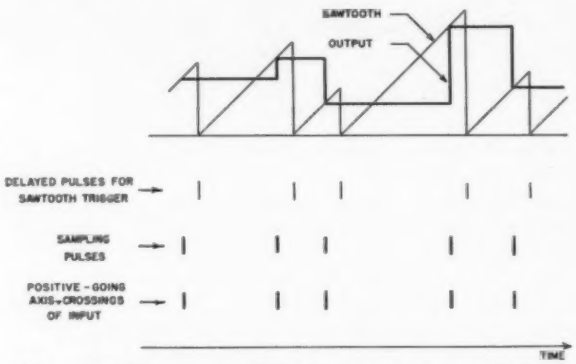

FIG. 2. Principle of operation of Axis-Crossing Interval Meter.

It can be shown that $c(\tau)$ and $C(f)$ are a Fourier transform pair. That is

$$C(f) = \int_{-\infty}^{\infty} c(\tau) \exp(-2\pi i f \tau)\,d\tau \qquad (4a)$$

and

$$c(\tau) = \int_{-\infty}^{\infty} C(f) \exp(2\pi i f \tau)\,df. \qquad (4b)$$

It is often convenient to normalize $c(\tau)$ by dividing all values by $c(0)$. The normalized autocorrelation function will be denoted $\rho(\tau)$. Since

$$c(0) = \int_{-\infty}^{\infty} C(f)\,df$$

from equation (4b), it is clear that if $C(f)$ has an "area" of unity, equation (4b) will yield $\rho(\tau)$ directly.

It has been pointed out that we will be dealing with randomly fluctuating signals, which are best described in terms of statistical characteristics. Such a signal is or may be considered to be a member of an infinite ensemble of signals having the same statistical properties (though not necessarily the same waveform, of course); all members of the ensemble are thought of as having been generated by the same random process. If the statistical properties are constant with time, the signals are called stationary. Two kinds of averages are employed in dealing with random signals. The first is a time average performed on a single member of the ensemble. The second is an ensemble average, performed over all members of the ensemble at a fixed instant of time. An ergodic process is a stationary process for which time averages and ensemble averages yield the same results. It is assumed that the random processes dealt with here are ergodic. Then we may write

$$\begin{aligned} c(\tau) &= \int_{-\infty}^{\infty} u(t)\, u(t-\tau)\, dt \\ &= \overline{u(t)\, u(t-\tau)} \end{aligned} \tag{5}$$

where a line over a quantity will be used to denote an ensemble average.

Mean-Square Deviation in Period

The ith positive-going axis-crossing of the input signal (pure tone of frequency f_o and period τ_o) to the recorder occurs at time t_i. Letting v_R denote the nominal recording speed, the position at which this axis-crossing is recorded on the tape is

$$\begin{aligned} x_i &= v_R t_i + \delta_{Ri} \\ &= v_R i \tau_o + \delta_{Ri}. \end{aligned} \tag{6}$$

Here, δ_{Ri} is the positional error introduced by noise in the record amplifier, irregular tape motion, and residual noise on the tape.

The playback time t_i' of the ith axis-crossing may be written

$$t_i' = \frac{x_i}{v_p} + \theta_{pi} \tag{7}$$

where v_p is the nominal playback speed, and θ_{pi} is the time error introduced by irregular tape motion and noise in the playback amplifier. Using equation (6), equation (7) becomes

$$t_i' = \frac{v_R}{v_p} t_i + \frac{\delta_{Ri}}{v_p} + \theta_{pi}.$$

This equation may be written in the simple form

$$t_i' = \frac{v_R}{v_p} t_i + \theta_i \tag{8}$$

where θ_i denotes the total time error introduced by the recording and playback processes. θ_i may be thought of as a sample, taken at time t_i', of a continuous fluctuating time error signal $\theta(t')$.

The interval between two successive positive-going axis-crossings upon playback is

$$\begin{aligned} \tau_i' &= t_i' - t_{i-1}' \\ &= \frac{v_R}{v_p}(t_i - t_{i-1}) + \theta_i - \theta_{i-1} \\ &= \frac{v_R}{v_p}\tau_o + \theta_i - \theta_{i-1}. \end{aligned} \tag{9}$$

Since it may be assumed that $\theta(t')$ has zero mean, the mean (ensemble average) expected axis-crossing interval upon playback is

$$\tau_o' = \overline{\tau_i'} = \frac{v_R}{v_p}\tau_o: \tag{10}$$

Then, the mean-square deviation in period may be written

$$\begin{aligned} \sigma^2 &= \overline{[\tau_i' - \tau_o']^2} \\ &= \overline{[\theta_i - \theta_{i-1}]^2} \\ &= 2\overline{\theta^2(t')} - 2\overline{\theta_i \theta_{i-1}}. \end{aligned}$$

θ_i and θ_{i-1} are separated, on the average, by the fixed interval τ_o'. Thus, from equation (5)

$$\begin{aligned} \sigma^2 &= 2\overline{\theta^2(t')} - 2c_\theta(\tau_o') \\ &= 2\overline{\theta^2(t')}\,[1 - \rho_\theta(\tau_o')]. \end{aligned} \tag{11}$$

Because the contributions from noise and from motional effects (flutter) to σ^2 are independent, the expression in equation (11) may be split into two terms

$$\sigma^2 = \sigma_n^2 + \sigma_m^2 \tag{12a}$$

where

$$\sigma_n^2 = 2\overline{\theta_n^2(t')}\,[1 - \rho_{\theta n}(\tau_o')] \tag{12b}$$

and

$$\sigma_m^2 = 2\overline{\theta_m^2(t')}\,[1 - \rho_{\theta m}(\tau_o')]. \tag{12c}$$

The quantities in equation (12) will be considered in the next two sections.

Mean-Square Deviation Due to Noise

Some comments concerning noise are pertinent at this point. Two types of noise may be expected in the played-back signal. The first is the "noise-behind-the-signal" or "modulation noise" encountered in recording with AC bias.[4] The amplitude of modulation noise is approximately proportional to signal amplitude and will be considered negligible near the axis-crossing. The second type of noise is that added to the signal. In this case, the noise amplitude is essentially independent of signal amplitude and need not be small near axis-crossings; therefore, this type of noise is the more likely cause of significant time errors detected by the ACIM. It will be assumed that this noise is ergodic

[4] S. J. Begun, *Magnetic Recording*, (Murray Hill Books, New York, 1949), pp. 69-73.

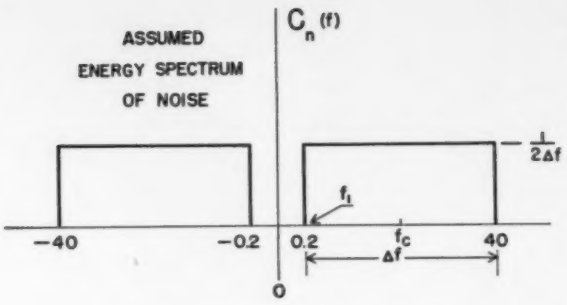

FIG. 3. Assumed energy spectrum of noise.

and has a flat energy spectrum distributed over the entire response range of the recorder.

Neglecting the effects of flutter, the amplitude of the signal plus noise at time t' during playback is

$$s(t') + n(t') = A \sin 2\pi f_o' t' + n(t')$$

where f_o' is the nominal frequency of the played-back signal

$$f_o' = \frac{1}{\tau_o'} = \frac{v_p}{v_R} f_o$$

and $n(t')$ represents the total noise introduced during recording and playback. At the axis-crossings, the times of which are given by equation (8), the signal becomes

$$\begin{aligned} s(t_i') &= A \sin 2\pi f_o' \left(\frac{v_R}{v_p} t_i + \theta_{ni} \right) \\ &= A \sin \frac{2\pi}{\tau_o'} (i\tau_o' + \theta_{ni}) \\ &= A \sin \left(2\pi i + \frac{2\pi\theta_{ni}}{\tau_o'} \right) \\ &= A \sin \frac{2\pi\theta_{ni}}{\tau_o'}. \end{aligned} \tag{13}$$

If the signal-to-noise ratio is sufficiently large, θ_{ni} is small compared to τ_o' and equation (13) may be written

$$s(t_i') \simeq \frac{2\pi A}{\tau_o'} \theta_{ni}. \tag{14}$$

At the axis-crossings, we have the condition

$$s(t_i') + n(t_i') = 0.$$

From equation (14), this condition may be written in the form

$$\frac{2\pi A}{\tau_o'} \theta_{ni} + n_i = 0 \tag{15}$$

where n_i is used as an abbreviation for $n(t_i')$. Similarly, at the preceding axis-crossing

$$\frac{2\pi A}{\tau_o'} \theta_{n,i-1} + n_{i-1} = 0. \tag{16}$$

From equations (9), (15), and (16), the deviation in period is

$$\begin{aligned} \tau_i' - \tau_o' &= \theta_{ni} - \theta_{n,i-1} \\ &= \frac{\tau_o'}{2\pi A} (n_{i-1} - n_i). \end{aligned}$$

By the same reasoning used to obtain equation (11), the mean-square deviation in period caused by noise may be written

$$\sigma_n^2 = \left(\frac{\tau_o'}{2\pi A} \right)^2 \cdot 2\overline{n^2(t')} \, [1 - \rho_n(\tau_o')]$$

where $\rho_n(\tau)$ is the normalized autocorrelation function of the noise itself. If the rms signal level S_o and rms noise level N_o are introduced, σ_n^2 becomes

$$\sigma_n^2 = \left[\frac{\tau_o'}{2\pi (S_o/N_o)} \right]^2 \cdot [1 - \rho_n(\tau_o')]. \tag{17}$$

The form of $\rho_n(\tau)$ may be obtained explicitly by taking the inverse Fourier transform of the energy spectrum of the noise. If a rectangular spectrum is assumed for the noise, this calculation yields

$$\rho_n(\tau) = \frac{\sin \pi(\Delta f)\tau}{\pi(\Delta f)\tau} \cos 2\pi f_c \tau \tag{18}$$

where

Δf = noise bandwidth

and

f_c = center frequency of positive noise band.

The actual spectrum chosen is shown in Figure 3, and the autocorrelation function is plotted in Figure 4. The fact that the (positive) noise band has a lowest frequency of 200 cps. instead of zero has negligible effect upon the autocorrelation function. This may be verified by noting that

$$f_c = \frac{\Delta f + 2f_l}{2}$$

where f_l is the lowest frequency in the positive noise band,

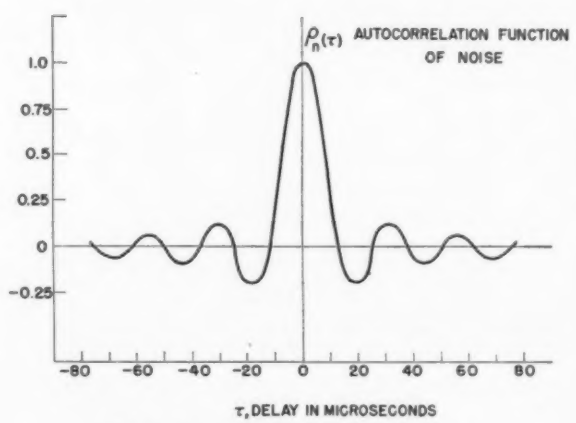

FIG. 4. Autocorrelation function of noise.

and that in this case f_1 is small compared to Δf. The zeros of $\rho_n(\tau)$ are determined by both of its factors: the zeros of $\sin \pi(\Delta f)\tau / \pi(\Delta f)\tau$ occur at integral multiples of $1/\Delta f$, or at integral multiples of 25 microseconds; the zeros of $\cos 2\pi f_c \tau$ occur at odd multiples of $f_c/4 \approx 1/2\Delta f$, or at odd multiples of about 12.5 microseconds.

Mean-Square Deviation in Period Due to Flutter

That the primary motional effects arise from vibrations has been demonstrated by several independent studies.[1,5] Ordinarily, a single high-Q vibration is involved, and it has been generally agreed to be a longitudinal vibration in the tape; this is the situation to be treated here.

It should be pointed out that, in addition to such systematic motional effects as vibrations, random irregularities in tape motion and contact with the record and playback heads are known to be present. However, such effects will be treated as giving rise to random noise, and the resulting time errors are assumed to have been handled in the previous section.

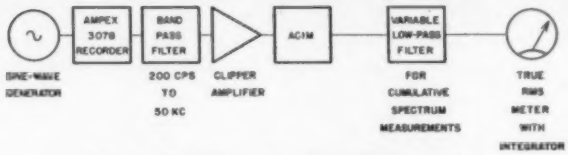

FIG. 5. System for time error measurements.

The energy spectrum of the vibration, and thus of the error signal $\theta_m(t')$ is considered to be contained in a narrow band about a central frequency f_v. A convenient form to assume for this spectrum is

$$C_{\theta m}(f) = \frac{a}{2\pi}\left[\frac{1}{a^2 + (f - f_v)^2} + \frac{1}{a^2 + (f + f_v)^2}\right]. \quad (19)$$

Here, $2a$ is the vibration bandwidth and a is small compared to f_v. Spectrum measurements,[1] made with the system in Figure 5, indicated that f_v is approximately 2.5 kc and $2a$ is approximately 300 cps. The half of $C_{\theta m}(f)$ which is centered at $+f_v$ is shown in Figure 6 for these values of f_v and $2a$.

The inverse transform of equation (19) yields the desired result

$$\rho_{\theta m}(\tau) = \exp(-2\pi a|\tau|)\cos 2\pi f_v \tau. \quad (20)$$

Thus, the autocorrelation function of the time errors due to flutter is a cosine of frequency f_v, with an exponential "envelope" whose rate of decay is determined by the vibration bandwidth. This function is plotted as the solid curve in Figure 7, for the values of f_v and $2a$ used in Figure 6.

To summarize, the mean-square deviation in period may be written in the form

[5] R. A. Von Behren and R. J. Youngquist, "Frequency-Modulation Noise in Magnetic Recording," *J. Audio Eng. Soc.*, 3, 26 (1955).

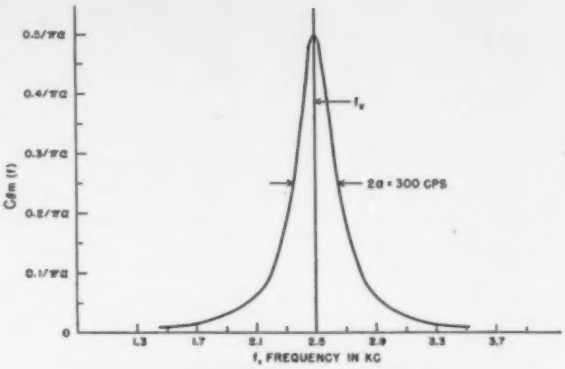

FIG. 6. Assumed energy spectrum of flutter.

$$\sigma^2 = \left[\frac{\tau_o'}{2\pi(S_o/N_o)}\right]^2 \cdot [1 - \rho_n(\tau_o')] + 2\overline{\theta_m^2(t')}\,[1 - \rho_{\theta m}(\tau_o')]. \quad (21)$$

The autocorrelation functions are given by equations (18) and (20), respectively.

Autocorrelation Function of ACIM Output

The above results have been in terms of the autocorrelation functions $\rho_n(\tau)$ and $\rho_{\theta m}(\tau)$. Since the autocorrelation function actually to be measured is $\rho_z(\tau)$, that of the ACIM output, the interrelationships are desired.

Assuming noise effects to be absent, we relate $\rho_{\theta m}(\tau)$ and $\rho_z(\tau)$. $\rho_z(\tau)$ may be evaluated as follows. Consider the two axis-crossing intervals on playback

$$\begin{cases} t_o' \text{ to } t_1' \\ t_k' \text{ to } t_{k+1}' \end{cases}$$

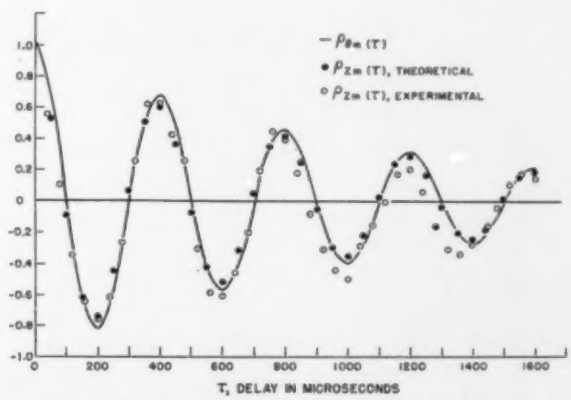

FIG. 7. Autocorrelation functions associated with flutter.

From equation (8), these intervals may be written

$$\begin{cases} \left(\frac{v_R}{v_p}t_o + \theta_{mo}\right) \text{ to } \left(\frac{v_R}{v_p}t_1 + \theta_{m1}\right) \\ \left(\frac{v_R}{v_p}t_k + \theta_{mk}\right) \text{ to } \left(\frac{v_R}{v_p}t_{k+1} + \theta_{m,k+1}\right) \end{cases}$$

The total time error in the first interval is $(\theta_{m1} - \theta_{mo})$. The total time error in the second interval, which is separated from the first by about $k\tau_o'$, is $(\theta_{m,k+1} - \theta_{mk})$. The autocorrelation function of the ACIM output (see the description of the ACIM above) is

$$\begin{aligned} c_{zm}(k\tau_o') &= \overline{(\theta_{m,k+1} - \theta_{mk}) \cdot (\theta_{m1} - \theta_{mo})} \\ &= \overline{\theta_{m,k+1}\theta_{m1}} + \overline{\theta_{mk}\theta_{mo}} - \overline{\theta_{m,k+1}\theta_{mo}} - \overline{\theta_{mk}\theta_{m1}} \\ &= 2c_{\theta m}(k\tau_o') - c_{\theta m}[(k+1)\tau_o'] - c_{\theta m}[(k-1)\tau_o']. \end{aligned} \quad (22)$$

Equation (22) may be written in the normalized form

$$\rho_{zm}(k\tau_o') = \frac{2\rho_{\theta m}(k\tau_o') - \rho_{\theta m}[(k+1)\tau_o'] - \rho_{\theta m}[(k-1)\tau_o']}{2[1 - \rho_{\theta m}(\tau_o')]}. \quad (23)$$

It is seen that, because of the nature of the ACIM output, the theory predicts values of the autocorrelation function $\rho_{zm}(\tau)$ only at integral multiples of τ_o'. The index k may be any positive or negative integer, or zero.

A similar computation, neglecting the effects of flutter, yields an analogous relation between $\rho_n(\tau)$ and $\rho_z(\tau)$.

$$\rho_{zn}(k\tau_o') = \frac{2\rho_n(k\tau_o') - \rho_n[(k+1)\tau_o'] - \rho_n[(k-1)\tau_o']}{2[1 - \rho_n(\tau_o')]}. \quad (24)$$

The total autocorrelation function of the ACIM output may, of course, be written

$$\rho_z(k\tau_o') = \tfrac{1}{2}[\rho_{zm}(k\tau_o') + \rho_{zn}(k\tau_o')]. \quad (25)$$

However, the experiments to be described were designed to handle the two terms in equation (25) separately, and they will be treated separately in the following sections.

COMPUTATIONS AND EXPERIMENTAL RESULTS

Autocorrelation Function of ACIM Output

Equations (18), (20), (23), (24) and (25) provide the basis for predicting the autocorrelation function of the ACIM output. This function should, because of property (d) listed above for autocorrelation functions, separate the effects of noise and the effects of flutter.

Equations (18) and (24) may be used to predict $\rho_{zn}(\tau)$, considering the effects of noise alone. Experimentally, the following conditions were employed:

$$v_R = v_P = 30 \text{ ips}; \; f_o = 20 \text{ kc}; \; \tau_o = \tau_o' = 50 \; \mu\text{sec}.$$

Figure 4 indicates that $\rho_n(\tau_o)$ is zero for these conditions, so equation (24) takes the form

$$\rho_{zn}(k\tau_o) = \frac{2\rho_n(k\tau_o) - \rho_n[(k+1)\tau_o] - \rho_n[(k-1)\tau_o]}{2}. \quad (26)$$

Equations (18) (or Figure 4) and (26) yield the results in the table below.

TABLE I. Predicted values of $\rho_{zn}(k\tau_o)$ for $\tau_o = 50$ μsec.

k	$\rho_n(k\tau_o)$	$\rho_n[(k+1)\tau_o]$	$\rho_n[(k-1)\tau_o]$	$\rho_{zn}(k\tau_o)$
0	1	0	0	1
±1	0	0	1	−0.5
±2	0	0	0	0
±3	0	0	0	0

This function is plotted by the dark circles in Figure 8. Experimental points, obtained with the system of Figure 9, are indicated by the open circles in Figure 8. For these measurements, the recording was made at a low signal-to-noise ratio to emphasize the time errors due to noise. This technique is not entirely satisfactory, of course, but it is seen to yield results roughly in accord with those predicted in the table.

The shape of $\rho_{zn}(k\tau_o)$ has a simple physical significance. We are correlating the time errors in two axis-crossing intervals for various values of separation between the intervals. For $k = 0$, the two intervals are the same and the correlation between the errors must be unity. For $k = \pm 1$, the intervals are consecutive. Since consecutive intervals have an axis-crossing in common, a shift in the location of this axis-crossing increases the duration of one interval and decreases the length of the other; thus, the time errors in these intervals are negatively correlated. For larger magnitudes of k, the two axis-crossing intervals are separated by at least one interval and there is no correlation between the errors. The latter statement implies, of course, that τ_o

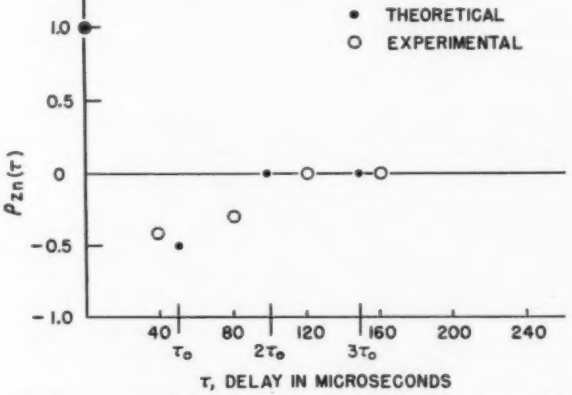

FIG. 8. Autocorrelation function of ACIM output for low signal-to-noise ratio.

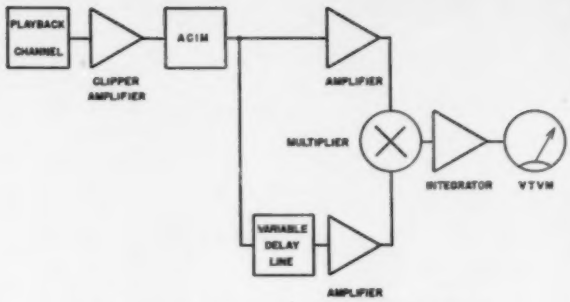

FIG. 9. System for autocorrelation measurements.

be larger than the "independence time" of the noise (i.e., larger than $1/\Delta f$ or 25 microseconds).

It is noted explicitly that the exact form of $\rho_{zn}(k\tau_0)$ depends on the choice of τ_0. Here, τ_0 was chosen such that all terms in (26) occurred at the central peak or at the zeros of $\rho_n(\tau)$. A larger value of τ_0 would result in about the same shape for $\rho_{zn}(k\tau_0)$. Smaller values of τ_0, in the vicinity of 25 microseconds, should produce a more oscillatory $\rho_{zn}(k\tau_0)$; however, the function should still damp out as τ approaches 100 microseconds.

Equations (20) and (23) may be used in a similar manner to predict $\rho_{zm}(\tau)$, considering the effects of flutter alone. For the experimental conditions indicated above, $\rho\theta_m(\tau_0)$ is 0.673, so (23) may be written

$$\rho_{zm}(k\tau_0) = \frac{2\,\rho_{\theta m}(k\tau_0) - \rho_{\theta m}[(k+1)\tau_0] - \rho_{\theta m}[(k-1)\tau_0]}{0.654}. \tag{27}$$

In this case, since τ_0 is considerably smaller than the period τ_v of the flutter,

$$\tau_v = \frac{1}{f_v} \approx 400 \text{ microseconds},$$

(27) should yield results which differ only slightly from the original function of (20). Points obtained using (27) are plotted as dark circles on Figure 7. A set of experimental points, obtained with the system in Fig. 9, are shown as open circles in Fig. 7. These measurements were made with a high signal-to-noise ratio to emphasize the effects of flutter. The effect of noise is still present, however, and produces a pronounced peak centered at $\tau = 0$. It may be seen from Figure 8 that the effects of noise should be negligible for values of τ greater than about 100 microseconds; therefore, the experimental data were adjusted to have the value predicted by (27) at the first negative peak ($\tau =$ 200 microseconds). For this condition, the spike at $\tau = 0$ has a value of about 1.9. Figure 7 indicates a fairly close agreement between the theoretical predictions made above and the experimental data.

As we predicted at the beginning of this section, autocorrelation analysis has achieved a separation of the broadband (large extent in frequency, small extent in τ) effects of noise and the narrowband (small extent in frequency, large extent in τ) effects of flutter. Autocorrelation data, such as those shown as open circles in Fig. 7, may thus be used to estimate the characteristics of the flutter. For this purpose, it will be assumed that τ_0 is sufficiently large compared to τ_0' so that (20) may be used directly as the theoretical form of $\rho_{zm}(\tau)$. The frequency of the vibration is given by the frequency of the cosine in the autocorrelation function, and is probably best measured by averaging the intervals between peaks in the data and taking the reciprocal of this average period. The vibration bandwidth is determined by the exponential decay of the autocorrelation function. At the positive peaks of the function, (20) becomes

$$\rho_{\theta m}(r\tau_v) = \exp(-2\pi a r \tau_v) \tag{28}$$

where r is a positive integer (not zero, because of the spike at $\tau = 0$) and only positive values of τ are being considered. Taking natural logarithms and employing the actual data points,

$$2a = -\frac{l_n\,\rho_{zm}(r\tau_0)}{\pi r \tau_v}. \tag{29}$$

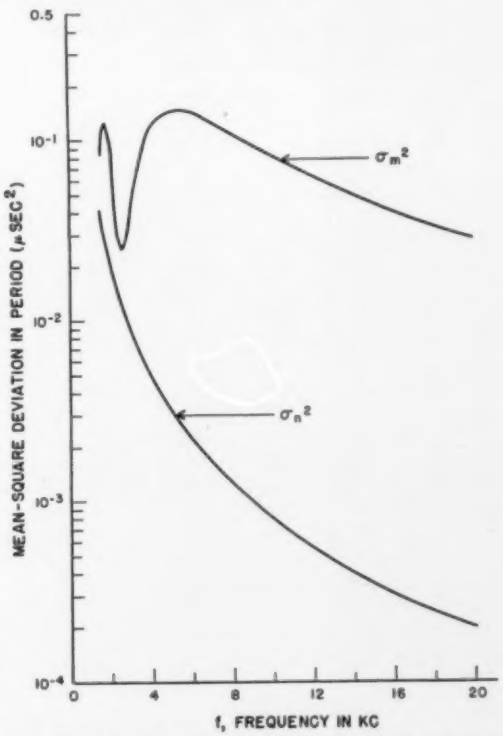

FIG. 10. Mean-square deviation in period *vs* recording frequency.

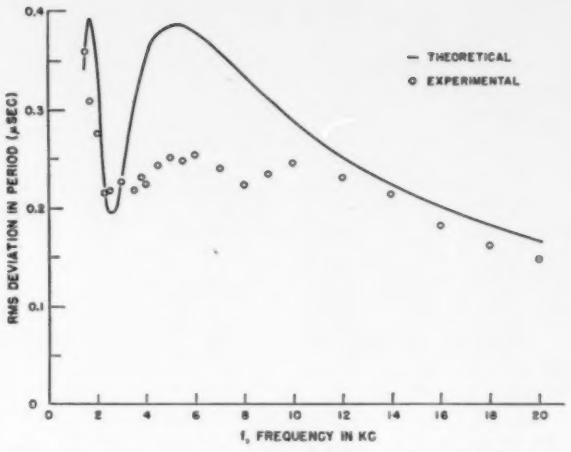

FIG. 11. Root-mean-square deviation in period *vs* recording frequency.

These results may be obtained with a fair degree of precision, and are preferable to direct spectrum measurements because of the more complete removal of the influence of noise.

Root-Mean-Square Deviation in Period

The quantity measured by the system in Fig. 5 is σ, the rms deviation in period. From a theoretical standpoint, this quantity is the square root of the expression in (21).

Equation (21) may be employed to predict the effect upon σ of varying recording frequency f_o. The following experimental conditions are assumed:

$$v_R = v_P = 30 \text{ ips};$$

$$\tau_o' = \tau_o = \frac{1}{f_o};$$

$$S_o/N_o \approx 55 \text{ db};$$

$$\overline{\theta_m^2(t')} \approx 4 \times 10^{-2} \ [\mu\text{sec}^2].$$

The signal-to-noise ratio of 55 db is that specified by Ampex for recording levels near saturation. The value of mean-square time error due to flutter was estimated from experimental data and corresponds to an rms positional error of about 6 microinches on the tape. A separate theoretical curve, based on these experimental conditions, for each term in (21) is presented in Fig. 10. In Fig. 11. predicted values of σ,

$$\sigma = (\sigma_n^2 + \sigma_m^2)^{1/2}$$

are plotted as the solid curve, and experimental data are shown as open circles. Except for the less pronounced peak at 5 kc, the experimental points agree fairly well with the theoretical values. Figure 11 indicates that the flutter effects should be taken into account in selecting a recording frequency. For example, recording frequencies in the vicinity of 2.5 kc and 5 kc do not appear advisable for the recorder under consideration here.

Similar methods may be employed to investigate the effects of variations in nominal recording and playback tape speeds. However, data were not taken for this purpose at the time of the other experiments reported; therefore, this topic will not be discussed here.

SUMMARY

Autocorrelation techniques have been employed to predict and measure the effects of time errors arising from random noise and flutter. These methods have been shown to yield theoretical and experimental results which are in fairly close agreement. The large degree of separation of noise and flutter effects achievable with autocorrelation makes this approach preferable to direct spectrum measurements.

ACKNOWLEDGMENTS

For the original concepts present in this paper and for guidance provided during the experiments, the author is grateful to F. Mansfield Young (presently with Adage, Inc., Cambridge, Mass.). Thanks are also due to Dr. E. C. Westerfield of the U. S. Navy Electronics Laboratory for his thorough critical review of the theoretical development.

THE AUTHOR

Roger H. Prager was born in New York City, November 9, 1931. He received the B.A. degree from the University of Chicago in 1951, and the B.S.E.E. degree from M.I.T. in 1955. He was for a year and a half a member of the research staff in the M.I.T. Acoustics Laboratory, engaged in research on the statistical analysis of sonar signals using analog computing techniques. Since January 1957 Mr. Prager has been an Electronic Scientist at the U. S. Navy Electronics Laboratory in San Diego. His work has been in the areas of sonar signal processing and display.

Endless Loop Magnetic Tape Tree*

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The introduction of four-track pre-recorded magnetic tape to be reproduced at a play speed of 3-3/4 ips permitted the use of several manufacturing techniques new to the recording industry. One of these techniques involved the use of an endless loop master tape. This paper describes the design and development of the "tape tree" used to transport the endless loop tape.

THE MAGNETIC tape cartridge, described by Mr. Andrews, presented new problems in the recording of pre-recorded tape.[1] These problems were due chiefly to two things: (1) the shorter length of tape required, since the tape runs at a speed of 3¾ inches per second instead of 7½ ips; and (2) the short leaders necessary at the beginning and end of the recording, which are desirable to prevent long pauses between musical selections or parts of one selection.

The technique used for recording tape records prior to the tape cartridge was to prepare a master tape from which exact duplicates are to be copied. To record these copies, the master tape is run through a tape transport which drives the tape past the magnetic heads from a supply reel to a take-up reel. These heads electrically transfer the music signal from the master tape to the record heads of the slave units. The record heads of the slave units place the signal on ¼ in. tape, which then is wound onto a new reel. This copy on its reel is removed for packaging and shipping, at which time the master tape is rewound, new reels placed on the slave units, the tape attached to these reels, and the record operation repeated.

Applying this same technique to tape for the tape cartridge requires each copy to be rewound to remove approximately 10 ft. of blank tape, the amount necessary to get the slave units running at constant speed. This wasted tape could be eliminated if the entire 5400 ft. supply reel of blank tape were recorded without stopping, leaving the desired leader length between each copy. To do this, either two masters running on synchronized transports or a continuous loop master would be required. The two-master system would present considerable added expense, and the automatic switching of the transports becomes very involved.

The use of a continuous master can be justified on the following bases:

(1) An increase in production due to the elimination of rewinding the master.
(2) A labor savings for cutting the leader to the correct length, since one cut will produce the end of one tape and the beginning of another.
(3) A savings in tape usage, since one start-up time is required for the slave units to produce several copies.

The continuous loop master tape was to be ½ in. wide and have a maximum length of 1200 ft. It was to be driven by the drive pulley of the transport past the magnetic heads at a speed of 30 ips. Several methods of transporting this tape were tried, but, due to the urgency of the project, most of the development time was spent on three methods. The methods will be called (1) the two-pulley method, (2) the multiple pulley method using conventional ball bearings, and (3) the multiple air-idler method.

TWO-PULLEY METHOD

The two-pulley method consists of winding the required length of tape onto two pulleys which are mounted a considerable distance apart.[2] A description of how the tape is loaded onto the two pulleys is as follows (see Fig. 1): From a supply reel of the tape transport, onto which can be placed a fixed drag, the tape is threaded through the tape transport to the top pulley. From there the tape is pulled over the top pulley, given a one-half twist, and then pulled down to the lower pulley. The tape is pulled around the lower pulley, given another one-half twist and returned to the top pulley. Here the end is tucked under the previous layer of tape so that the tape is wound onto the inside diameter of the top

* Revised copy received March 24, 1959.
† Design Engineer.
1. "High Quality Reproduction of Magnetic Tape in a Cartridge" by D. R. Andrews, presented at A.E.S. Convention October 2, 1958.
2. For suggesting this method we are indebted to Mr. W. R. Isom and Mr. H. R. Warren, of Defense Electronic Products, Camden, N. J.

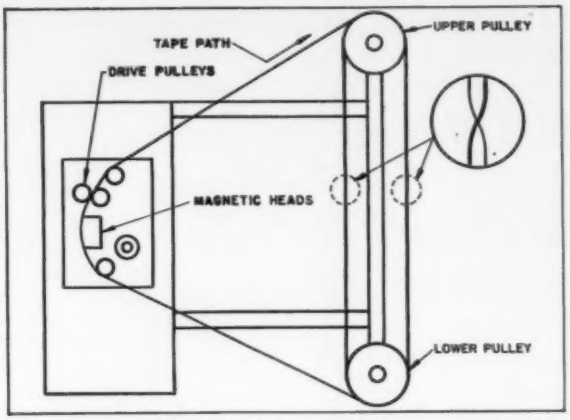

FIG. 1. Two pulley method for transporting a continuous loop magnetic tape.

pulley. This process is repeated until the desired length of tape has been wound between the two pulleys. To splice the tape ends and produce a continuous loop, the tape is taken from the bottom pulley and spliced to the tape end at or near the supply reel of the tape transport. The tape master is now ready to run continuously, being driven by the drive pulley of the tape transport from the bottom pulley, past the magnetic heads, and onto the top pulley.

For a given pulley size and loading pull, the two-pulley system has a maximum capacity. This is due to the change in velocity of the tape as it passes from the outside diameter of one pulley to the inside diameter of the other. The capacity is reached when the tape relaxes (reaches zero tension) in this strand. Since no slippage occurs between tape faces, this theory can be expressed by the following equation:

$$P = \frac{Et^2Wx}{r + xt}$$

Where P = Loading force on tape in lb
E = Tension modulus for tape in psi
W = Tape width in inches
t = Tape thickness in inches
r = Radius of pulleys in inches
x = Number of layers of tape on the pulleys

Using average tape dimensions, which were determined by tension tests, and a tape length of 1200 ft., the following systems would produce a satisfactory working unit:

Pulley diameter (in inches)	Distance between pulleys (in feet)	
	Mylar base tape	Acetate base tape
12	43	67
18	27	43
24	21	33
30	17	27
36	14	22

These working units are based on a loading pull in the tape of one pound, which will not permanently distort the tape.

A model which was built to demonstrate the two-pulley method consisted of two 12-in. diameter pulleys mounted vertically on 100-in. centers. From this model and the calculated working units, it would appear the chief advantage of this method would be the low initial cost. Disadvantages are the time consuming loading operation, space requirements since the working dimensions are too large to make vertical erection possible, and the destruction of the master tape (which has considerable value) in case of malfunction.

MULTIPLE PULLEY METHOD USING CONVENTIONAL BALL BEARING PULLEYS

This method of continuously transporting tape is very straightforward. The tape is threaded from the vertically mounted tape transport and woven up and down over a series of ball-bearing pulleys far enough apart to best utilize the ceiling height in the room. After passing over the pulleys, the tape is returned to the tape transport where a continuous loop is made by splicing the tape ends together. The tape is then ready to run continuously, driven by the tape transport and transported on the number of ball-bearing pulleys required to contain 1200 ft. of tape.

To maintain good contact between the tape and the magnetic heads of the transport, a tension in tape of 135 g is necessary. This force was also used as the maximum force in order to hold tape stretch and possible distortion to a minimum. To accomplish this it is necessary to keep the tension drop across each pulley as low as practicable. The tension in the tight side of the loop being 135 g, the tension in the loose side as the tape leaves the transport can be calculated from the equation:

$$F_n = K^n F_0$$

Where F_n = Resulting tension, in grams, in tape after passing over n pulleys (tight side tension)
F_0 = Initial tension in slack side of tape, in grams
n = Number of pulleys
K = Constant for the pulley (determined from bearing diameter, friction, and radial load) and varies with pulley diameter

It can be observed that this equation is valueless without some information about the bearings and pulleys to be used. This data was synthesized from a model (see Fig. 2) which contained twelve pulleys mounted on two pulley beams (six pulleys per beam), which could be separated a distance of nine feet or could be brought together to facilitate loading. The data synthesized from this model is shown on the graph (see Fig. 3) and indicates a value of $K = 1.025$. Using the available ceiling height and a tape length of 1200 ft., it was decided the multiplicity of pulleys or "tape tree" should contain 95 pulleys. From the equation it can be determined that

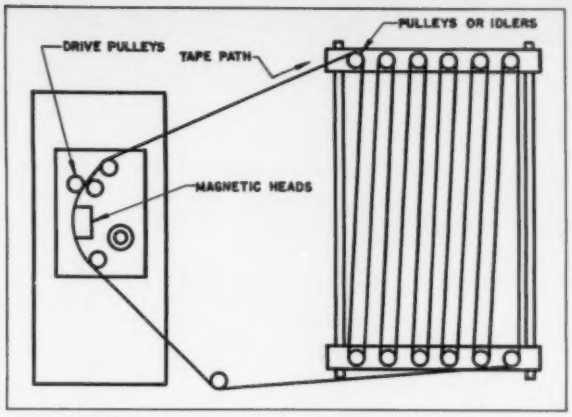

FIG. 2. Schematic of multiple pulley arrangement for transporting a continuous loop magnetic tape.

when F_n equals 135 g, the loose side tension becomes 13 g. Thirteen g in the loose side of the loop is not adequate tension to keep the tape on the pulleys. It was determined by experiment that at least 60 g tension is necessary in the loose side for tracking over the pulleys. Placing these requirements in the formula indicates that a value of $K = 1.010$ is required. If the same bearings are used, the pulley diameter necessary to satisfy this condition would be 4½ in.

From the model and calculations it would appear that this designed tape tree would be approximately 20 ft. long. Due to the sensitivity of the bearings, it would be expensive to make and maintain. The ease of loading this tree is very satisfactory.

MULTIPLE AIR-IDLER METHOD

An air-idler is a stationary flanged pulley over which the tape floats on a film of air (see Fig. 4). The air is supplied from small jets in the circumference of the idler. The method of transporting the tape on air-idlers is identical with the ball-bearing method. To synthesize the tree using air-idlers, the same model was used, replacing the ball-bearing pulleys with air-idlers.

The graph (see Fig. 5) represents the tape tension data that was synthesized from the model. The tensions were measured using air pressures over a range of 30 to 50 psi supplied to the idlers. This much change in pressure does not affect the operation of the tree. Starting with 135 grams tension in the tight side of the loop, the tape tension in the slack side drops to zero after passing over 95 idlers. The important consideration is that air-idlers need no tension in the tape to keep the tape on the idler. The tape will run smoothly and continuously over an air-idler with zero tension in the tape. The tension in the tight side of the loop has been increased to as much as 500 g with no difficulty encountered in the operation of the tree.

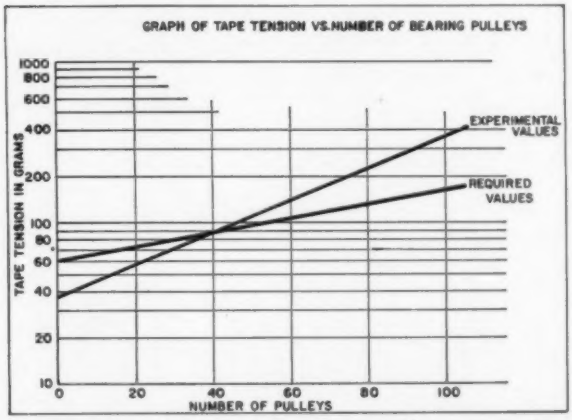

FIG. 3. Graph showing the effect of the number of conventional bearing pulleys on tape tension using a specific bearing.

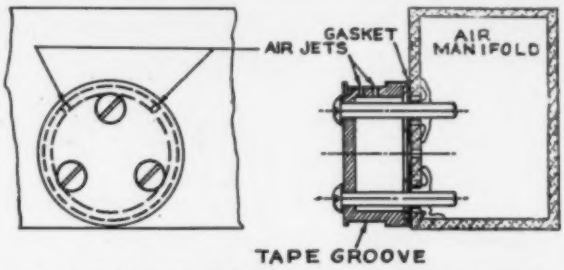

FIG. 4. Sketch of the air idler used to transport the magnetic tape master.

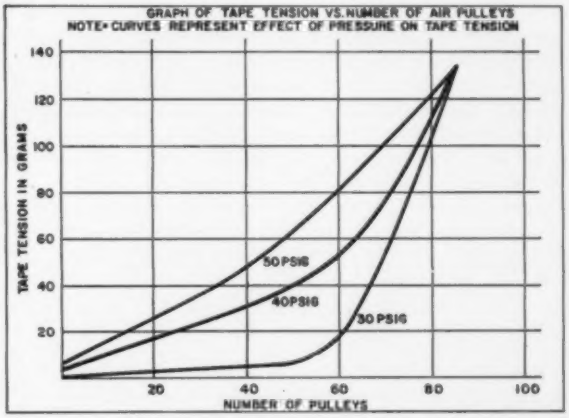

FIG. 5. Graph showing the effect of the number of air pulleys and the air pressure used on tape tension.

(*Continued on Page 96*)

A Single-Element Stereophonic Cartridge*

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The Westrex "45" stereo disc system is surveyed with particular emphasis on the design requirements for stereo cartridges. The principle of operation of a one-piece ceramic element, sensitive to displacement in two coordinates, is described. This element, a small cylinder with electrodes disposed about its inner and outer surfaces, is mounted in a plug-in type housing for ease of conversion and stylus replacement. Special damping and mounting methods are utilized to obtain proper values of vertical and lateral response and impedance. The paper is concluded with performance data on response, distortion, compliance and separation.

ALTHOUGH at the beginning it appeared that cost and space considerations might limit the enjoyment of stereophonic sound to the high-fidelity field, the stereo disc has at last brought this advance within the reach of everyone. Availabilitv of relatively inexpensive components and records has allowed manufacturers to cover the field from table units on up. The rapid development of the stereo-disc system from demonstrations in the fall of 1957 to the extensive lines now available has amazed even those of us who have been working on it over the period. Yet the quality of performance has become comparable to the high standards of monaural records and stereo tape. In this paper one link in this system, the phonograph cartridge, will be discussed, with particular emphasis on the requirements for stereo design.

Shortly after the revival of the Westrex 45/45 and the vertical-lateral systems, development of the Electro-Voice Stereo Cartridge began. Several design approaches were considered. It was soon apparent that several unique problems existed.

COMPLEX STYLUS MOTIONS

Briefly reviewing the Westrex 45/45 system will indicate the complex motions that can be expected of a stylus tracing a stereo groove, and therein lies the first problem. These motions considerably aggravate the tracking requirements of a cartridge.

In the low-frequency range, in which microphone spacing is small with respect to the wavelength, a lateral groove will result. This phasing was agreed upon to facilitate tracking of the large-amplitude, low-frequency signals, and is the resultant of an upward 45-degree motion of one channel and a downward 45-degree motion of the other channel. In true stereophonic recording, an exact 45-degree motion is very unlikely as some signal will appear in both channels.

Simultaneous acoustic pickup of sound by two or more microphones is an important requirement for the stereo illusion, which is dependent on phase and stochastic characteristics of the signal in addition to mere localization of the sound source from one speaker or the other by the directional properties of the listeners' hearing. It must be admitted, however, that a great many recordings are made with complete isolation between channels initially assured by placing the various instruments or sound sources in different studios or by recording them at different times. This practice, of course, has its place for dramatic and special effects, and is popular for demonstrations of the new system. In this situation exact 45-degree motion can occur, with a signal in one channel only. In the great majority of situations, however, vectors of motion will occur at all angles, and stylus motion and the electrical output of the transducers must correspond accurately. This motion can be displayed by connecting the cartridge channels to the vertical and horizontal inputs of an oscilloscope.

Accordingly, equal vertical and lateral compliances are indicated to prevent excessive forces on the stylus and groove wall with resulting distortion and record damage.

With lateral modulation, some vertical stiffness is necessary to overcome the vertical components (due to the 45-degree groove walls) of the acceleration and displacement forces. In monaural designs a ratio of lateral to vertical compliance of about two is considered desirable to avoid tracking problems. However, a ratio of one, as required for stereo designs, places a severe limitation on effective mass,

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† Chief Engineer, Cartridges.

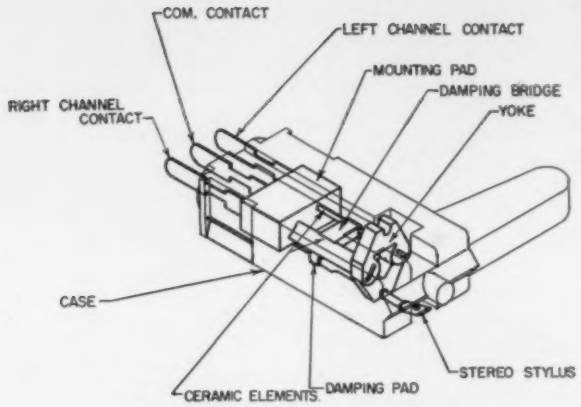


FIG. 1. A dual element stereo cartridge.

damping, and the compliance figure. Experience with stereo cartridges in various tone arms, on both stereo and monaural discs, has indicated a practical compliance of about 1. x 10^{-6} cm/dyne or higher.

CHANNEL ISOLATION AND CROSS-TALK

A second problem is that of obtaining adequate isolation or channel separation in a stereo cartridge. It has become customary to call isolation, or separation, the ratio of signal from the desired channel to the signal from that channel produced by mechanical drive in the undesired direction.

Cross-talk is considered as the ratio of signal from the desired channel to the signal from the undesired channel with a given mechanical drive. An isolation figure does not involve the sensitivity of either channel, while interpretation of cross-talk requires allowance for individual channel sensitivities.

At mid-range frequencies (below 4 kc for example), isolation can be obtained by straightforward mechanical means. A rather simple mechanical linkage of the yoke and the two ceramics is one way of isolating one coordinate of motion from the other. Secondly, element mounting can be arranged to provide greater compliance in the direction in which output is not desired. Of course, the inherent sensitivity pattern of the ceramic elements is another important means of attaining isolation. The sensitivity of these elements is a cosine function (with the angle measured about their axis), and therefore forces applied in the plane of the ceramics will produce no appreciable output. This is due to electrical cancellation of the two portions of the bender type ceramic.

In this frequency range, separation figures of 25 db are not difficult to obtain. However, at the higher frequencies from 4 kilocycles and up, a second problem, that of resonances in the element and the coupling system, becomes more evident. In a poor design, it is entirely possible for the polar pickup pattern of the cartridge to rotate 90 or even 180 degrees at certain frequencies, producing a higher cross-talk signal than that from the desired channel. While this crossover signal may occur over a relatively narrow frequency range as it is a resonance effect, the result can be unpleasant if the particular source material has predominant signals in this band. A high degree of distributed damping is required to control these resonances.

THE DUAL-ELEMENT CARTRIDGE

In Fig. 1, the dual element cartridge is shown in a phantom view. Two lead zirconia titanate (PZT) multimorph elements are coupled to a stylus by a resilient yoke which, through compliance in shear, decouples one channel from the other. The yoke also provides a degree of damping to the entire system.

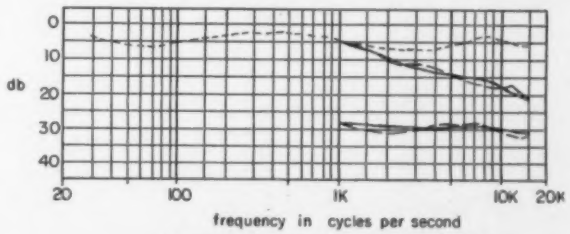
A second damping member bridges the ceramics to prevent half-wavelength resonance with a resulting loss of separation. Externally replaceable needles are clipped onto the case with the upper needle, with a 3-mil radius, positioned to couple to an upward projection of the drive yoke. The increased compliance and the geometry of this yoke projection compensate for the difference in recorded level between standard and LP records.

Performance figures for this cartridge are approximately as follows:

Output, .5 volt at 1 kc at 5½ cm/sec;
Lateral and vertical compliance, 2 x 10^{-6} centimeters per dyne;
Response, 20 to 20,000 cps (Fig. 2);
Channel separation, 25 db at 1 kc, 18 db at 5 kc and 12 db at 10 kc.

THE SINGLE-ELEMENT CARTRIDGE

For several years monaural plug-in type cartridges have been produced in several forms. One of these designs, known as the Power Point, has been widely used, and it was felt that a stereo version of this design would be very desirable. However, as the entire cartridge is less than ¼ in. in diameter and ¾ in. long, it did not seem feasible to include



------- frequency response (corrected to RIAA)
——— channel 1 } Westrex IA Record
— — — channel 2 }

FIG. 2. Frequency response and channel isolation.

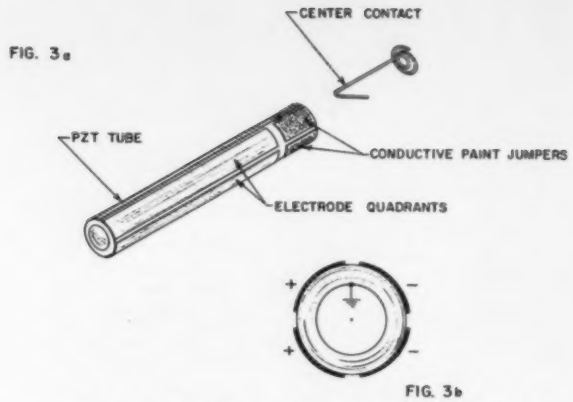


FIG. 3. A cylindrical two-channel transducer element.

a two-element mechanism in such a small package without sacrifice of some of the basic performance requirements. Accordingly, other configurations suitable for a stereophonic cartridge were investigated.

One of the most promising element forms, particularly from the standpoint of simplicity of fabrication and assembly, was the cylinder. In Fig. 3a is shown the construction of the cylindrical element. A lead zirconia titanate (PZT) tube approximately .050 in. in diameter by ½ in. long is electroded on the inner and outer surfaces with conductive silver. The outer electrode surface is divided into four quadrants and the element polarized in the manner shown in Fig. 3b. For improved symmetry, quadrants are individually polarized. Simultaneous polarization will produce unequal fringing fields between quadrants.

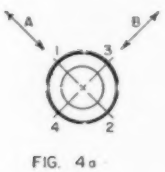
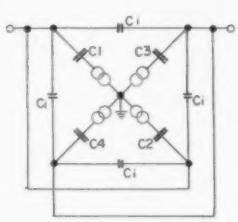
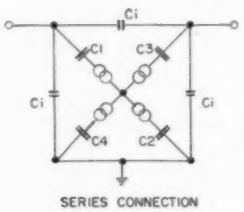


FIG. 4a



PARALLEL CONNECTION

FIG. 4b

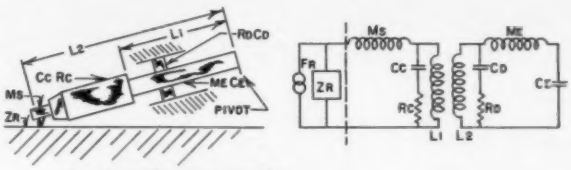


SERIES CONNECTION

FIG. 4c

FIG. 4. Electrical circuits of the cylindrical element.

To obtain maximum electrical capacity the element may be connected in parallel, with opposite diagonal quadrants bridged. To obtain maximum interelectrode shielding, it is desirable to ground the center electrode. Series operation of this element is possible for higher output, but isolation is limited to about 12 db by the common floating center electrode (Fig. 4c). With proper polarization a force *A* applied so as to produce a bend about the axes *1, 2* (Fig. 4a) will stress one quadrant in tension and the opposite in compression. Conversely, a force *B* applied so as to produce a bend about the axes *3, 4* will stress the other quadrants of the element, generating a potential with respect to ground. In each situation the electrodes adjacent to the neutral plane will be stressed in the same direction, and the generated potential will be cancelled. The degree of separation between



C_C—Compliance of coupler
C_E—Compliance of element
C_D—Compliance of damping (fulcrum) pad
M_E—Mass of element
M_S—Mass of stylus and mounting
R_C—Resistance of coupler
R_D—Resistance of damping pad
Z_R—Z of Record

FIG. 5. Approximate equivalent circuit of the single element cartridge.

the two channels obtainable in this element is affected by the small interquadrant capacities C_2 (Fig. 4b), the symmetry of the electrode surfaces, and the uniformity of the ceramic tube at frequencies below the primary resonance of the element. The small interquadrant capacities C_2 have values of about 18 $\mu\mu$f compared to 400 $\mu\mu$f per quadrant.

As slight asymmetries exist in a practical assembly, resonant breakup of the element may occur in planes other than the desired drive direction. This mode of vibration may, in fact, become helical about the axis at frequencies at which the element length is a multiple of a quarter wavelength. This element resonance must be controlled by a high degree of damping at the driven end. The usual methods of damping, between the moving element and the fixed walls of the housing can not be applied at this point without excessive loss of compliance. The better method of obtaining damping is through the parallel compliance C_c and resistance R_c of the coupling device between the stylus and element, as shown in the equivalent circuit of Fig. 5. This will increase the overall compliance of the system and, at the same time, decouple the element mass from the stylus thus lowering driving point impedance. Low stylus impedance is especially important in a stereo cartridge as the stereo groove is not essentially push-pull as is the monaural groove.

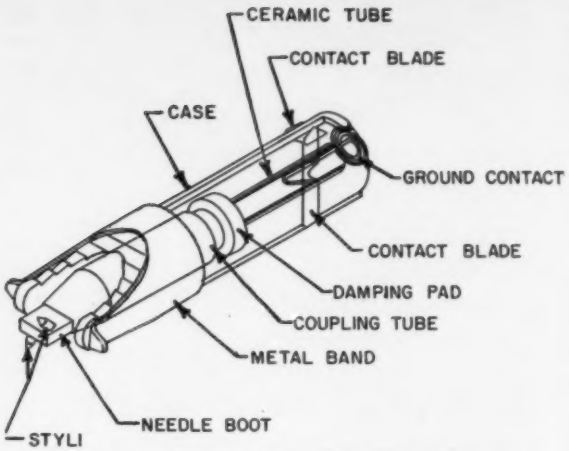


FIG. 6. Phantom view of the single-element cartridge.

The equivalent circuit of Fig. 5 indicates a second means of obtaining low stylus impedance. The lever action of the approximate dimensions L_1 and L_2 is represented by a transformer matching the element and mounting impedance on the right side to the desired low driving point impedance on the left.

In Fig. 6, a phantom view of a complete cartridge is shown. The center ground contact to the element is made through a spring clip inserted into the center hole. Contact to the two channels is made at the back by two metal blades extending through the housing. Each blade is notched to contact only the upper adjacent electrodes as the lower electrodes have been removed in this region. These blades also form a hinge support for the element. A fulcrum pad is positioned in the center of the case to support and center the entire assembly, and provide damping to the ceramic.

The driven end of the ceramic is coupled to the stylus cap by an elastic sleeve. This sleeve not only provides the required damping between the tube and the stylus but also largely determines the compliance-voltage product of the cartridge. The stylus cap bears two styli mounted as close to the axis of the element as possible to prevent torsional break-up of the system. The cap is made of a hard plastic in order to minimize drag or tangential motion of the stylus with the groove, and to provide a secure mounting for the styli. Angular orientation of the entire assembly within the case must be held to close tolerances as channel separation is inversely proportional to the sine of angular error. A polar plot of this cartridge is shown in Fig. 7 to illustrate the cosine form and the importance of accurate alignment of the nulls with the record. From this it can be seen that misalignment of the cartridge with the record by only a few degrees can cause a serious loss of separation.

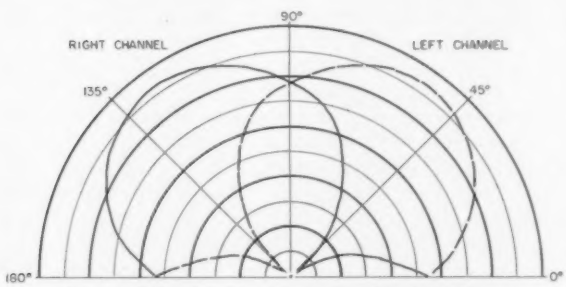


FIG. 7. Polar plot of 1 kc response (vertical plane).

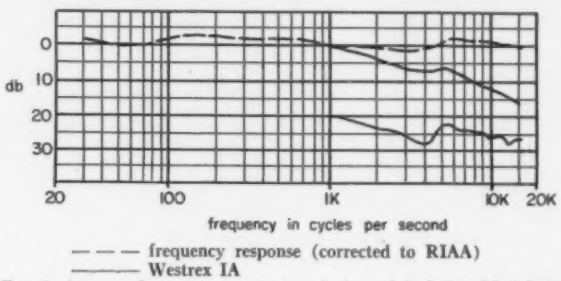


FIG. 8. Average frequency response and channel isolation Model 66.

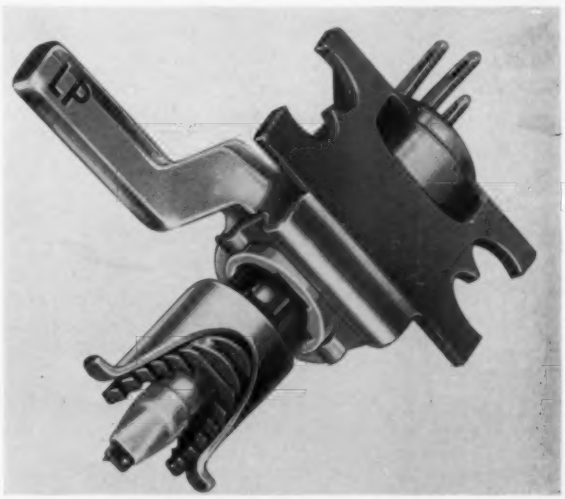


FIG. 9. The Model 66 Single Element Stereophonic Cartridge.

The housing end supports a pressed-on metal band to provide stiffness and mass to the housing. Spurious resonances of the housing are detrimental to separation, frequency response, and "needle talk." It was also necessary to redesign the mounting mechanism to reduce resonance trouble.

The mechanism shown is free of resonances and is also more simple than the original mechanism designed for monaural application.

PERFORMANCE OF THE SINGLE-ELEMENT CARTRIDGE

Fig. 8 illustrates typical response and separation characteristics for this cartridge. The low-end response, of course, depends on load resistance as this is a capacitive device. Since this capacity is approximately 800 $\mu\mu$f, a resistance of 3 megohm or higher is recommended for flat RIAA response below 30 cps. An average compliance of 1.5×10^{-6} cm/dyne combined with the .6 volt output (at 5.5 cm/sec at 1 kc) yields a voltage-compliance product of $.9 \times 10^{-6}$ (slightly higher than the usual range of dual element cartridges).

Distortion measurements made on records with velocities up to 18 cm/sec indicate excellent tracking capabilities throughout the reproduced frequency range.

ACKNOWLEDGMENTS

The author wishes to credit those who contributed in many ways to the success of this project. Among others, they are A. M. Wiggins, Erwin Dreiske and Dan Tomcik of the Electro-Voice Engineering Department, and the Clevite-Brush Electronics group for the development and production of the PZT tubes.

THE AUTHOR

John F. Wood began his studies in electronics with the U. S. Navy during World War II. He then attended the University of Illinois and received his B.S. in Electrical Engineering in 1950. He holds seven patents in the audio field. His professional affiliations are with IRE, PGA, and the Acoustical Society of America.

(*Continued from Page 91*)

A tape tree based on the performance of this model was designed and built to dimensions suitable for location in the New York Recording Studios (see Fig. 6). The tree was 9 ft. total length and 13 ft. 10 in. total height. It consisted of 95 air-idlers with a maximum distance between idlers of 158 in. or a tape capacity of $158 \times 95 \div 12 = 1250$ ft. when fully loaded. The idlers are mounted on hollow rectangular beams, which serve as the air manifold. These beams are guided on four 1-in. diameter rods, and each beam is raised and lowered by two tandem-operated screws driven by motors. To load the tree the beams are brought together a distance of 57 in. above the floor. At this point the tape is threaded from the transport through the idlers and back to the transport over two directional idlers. With the operator holding the end of the tape, the upper beam is then raised and the lower beam lowered to accommodate the length of the master tape. The two ends are then spliced together to form an endless loop. The loop passes over a counterbalanced pin at the last directional idler to indicate the proper running tension of 135 g.

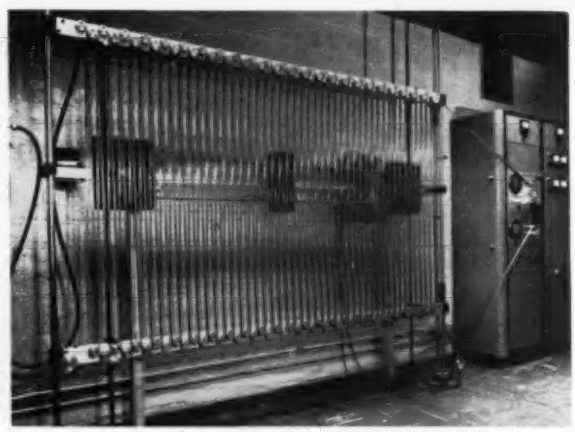


FIG. 6. Photograph of the completed "tape tree" installation in the RCA Recording Studio, New York.

After the tree had been installed and trial runs were in progress, it was discovered that a static problem existed when the tree was loaded with more than 800 ft. of one particular kind of tape. This static condition caused trouble when the tape was attracted to the tree or transport frames. This attraction caused the tape to drag against these frames and the entire loop to have a non-uniform motion. The static condition was reduced to a tolerable level by the addition of a "corona discharge" unit placed over the tape as it left the transport. One additional unit was placed in the low-tension segment of the tree. With this addition the tree was now ready to continuously run the tape master to make duplicate tapes for the magnetic tape cartridge.

Standards: Stepchild in the Laboratory*

C. E. WHITE†

AVCO R. A. D., 201 Lowell Street, Wilmington, Mass.

Significant psychoacoustic reactions: high-fidelity equipment performance measurements, loudspeaker acoustic responses—all are "authoritatively" expressed by measurements indicating high resolution and acurracy. Is the measuring equipment capable of supporting the conclusions drawn?

INTRODUCTION

OF PRIME importance in the recognition of inherent instrument error is the fact that scales of most ac meters are calibrated in terms of the rms value of a sine wave input. Any deviation of the input signal from this ideal will lead to erroneous indications. Furthermore, the response of the meter, whether thermal, peak indicating, average, or effective response will not alter this dependency upon ideal wave shape. It is established(1,2) that meter error is proportional to harmonics contained within the signal applied. The order of the harmonic and its phase relationship with the fundamental contribute to the error in the manner shown in Fig. 1. It is to be noted that an average-reading meter consistently reads low in the presence of complex waves. As the complexity of the signal changes by the addition of higher and higher order of harmonics, it has been shown(2) that the meter error approaches that for gaussian noise. For the average-reading meter the correct value is $v_{ind}/_{0.886}$.

The preceding deals with the accuracy of instrumentation. For the record, accuracy here is defined as the degree of correctness with which a measured value agrees with the true value which is established by *standards* of various types. All instrumentation is subject to internal constrictions of accuracy which are continuously variable over a period of time. These might be caused by bearing stickiness, temperature effect, zero drift, coil spring hysteresis, bearing wear, contact resistance, thermal voltages, external magnetic fields, eddy currents, etc.‡ For these reasons and others, it is well to bear in mind that a *precise* reading taken on a sensitive measuring instrument may not always be an *accurate* measurement.

VARIABILITY OF ERRORS

Operator's errors or the "human element" are affected by, or dependent upon, training, skill, sense of precision appreciation, complacent or opinionated attitudes towards personal accuracy achievements, ability to select high-quality measuring instruments and standards with required precision capabilities, and an appreciation of the scope of accuracy evaluation. The last involves such factors as environment, understanding the importance of system error or malfunctioning as compared to instrument error, and correct application of calibration data. The term precision is most often defined as the repeatability of a measurement. Over a long period of time and with an experienced operator, a measurement might be repeatable to an extent such as to give an infinitesimal deviation. But at the conclusion of these measurements, no one could say that this optimum value was an accurate one. For nowhere in the series of measurements was there established an ability to relate the readings obtained to the true value. It should be borne in mind that "absolute certainty is the privilege of uneducated minds"(5).

There exists a tendency to correlate sensitivity of instrumentation with precision. This is a very dangerous correlation. For example, we have an instrument with a full scale reading of 100 v, and the scale is divided into 100 parts. Normal estimation of scale division spacing permits one-tenth increments to be interpolated. Theoretically, then, it is possible to make a reading of 0.1 v. But the slightest acquaintance with meters should alert the operator to the fact that the precision of measurement at this low point of the scale certainly does not approach the precision of a reading

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† Section Chief, Laboratory Services.

‡ Systematic errors of this type may be calculated by means of the logarithmic differential. This is based upon the assumption that the errors are small; therefore the same calculations may be applied to them as are applied to differentials(8). It is not proposed to enter into a discussion of this method. Anyone interested in absolute determination may obtain the information from text books or by consulting with their associated Standards Laboratory.

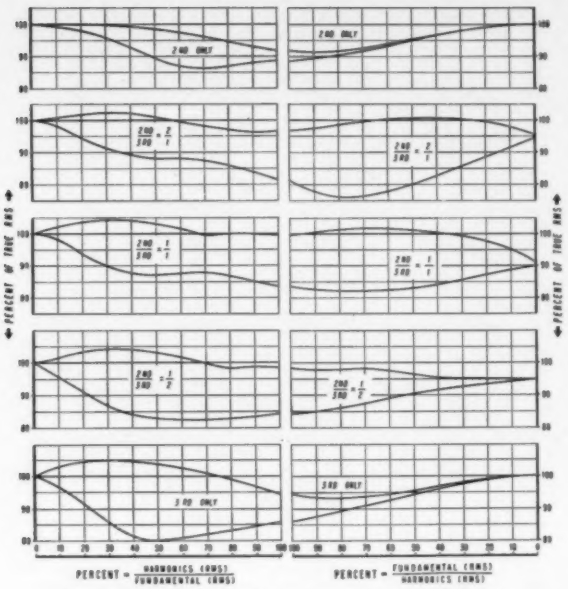


FIG. 1. Effects of harmonics on average-type voltmeter.

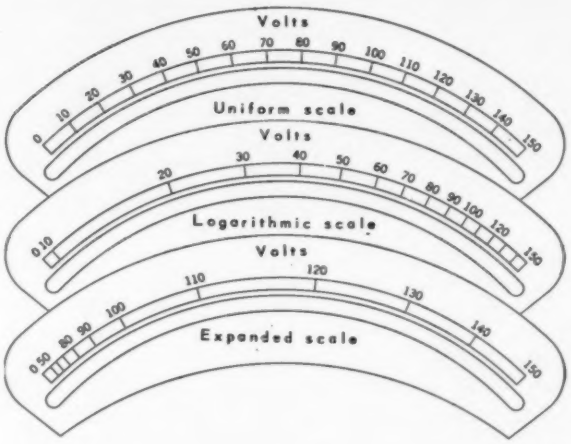
at full scale. This example is, of course, deliberately obvious. Unfortunately, the laboratory worker who makes readings at the half-scale point and thoughtlessly believes accuracy of the reading to be equal to that of the full-scale point forgets the fine print in the manufacturer's specification which stated "accuracy is x percent at full-scale reading." If it is at all necessary to retain accuracy over a great part of the scale, use should be made of a uniform scale such as is shown at the top of Fig. 2, which demonstrates superiority in readability over a greater scale range than the logarithmic or expanded scale.

An observer, in reading a meter deflection, is a victim of numerous physical phenomena. Among these are parallax, variability of estimation of pointer position, thickness of line width, etc.

Parallax is compensated for on precision instrumentation by inclusion of a mirror in the plane of the indicating scale. Use of a single line of sight, permitting coincidence of pointer and reflected image, will establish the correct line of sight to the scale.

The subject of pointer position estimation has been covered by Bäckström in great detail(6). An average operator estimates the position of the meter pointer to one-tenth of a scale division. The accuracy of such an estimation has been determined subjectively and is shown in Fig. 3. Bäckström determined that the various tenth divisions were not estimated with the same ability by his subjects, but individual tenths were selected with frequencies which were repeatable for single subjects over a long period of time. Also, the estimations were symmetrical about the midpoint of the division. A graph of a typical subject's ability to estimate is shown in Fig. 3a. Bäckström also brought out the fact that the shape of the graph was similar for various subjects although the actual figures differed. The variation of estimation of the physical spacing of the tenth-divisions for three observers is illustrated in Fig. 3b. It should be noted that if the position of the meter needle had been correctly estimated, all of the shaded blocks would have been of equal width and centered on the correct tenth-division Such an ideal case is shown at the bottom of the figure.

Another important contribution of Bäckström was his study of the influence of the thickness of the scale division lines upon estimation of the tenth-divisions. His results, shown graphically in Fig. 4, indicate an optimum choice of line width equivalent to one-tenth of the width of the division space. As a logical follow-up to this experiment, he further determined that the width of the scale divisions also affected determination of tenth-divisions. It was determined that the average error of observer estimation dropped



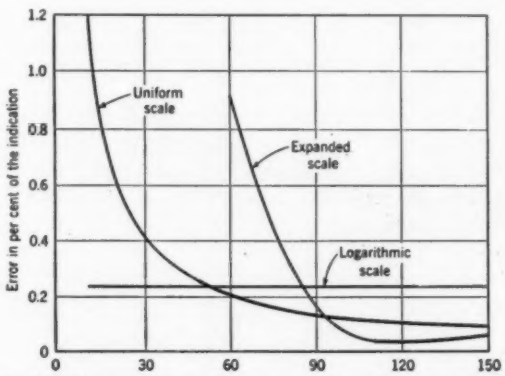


FIG. 2. Observational error in percent of reading caused by error of 5 mils on 6-inch scale.

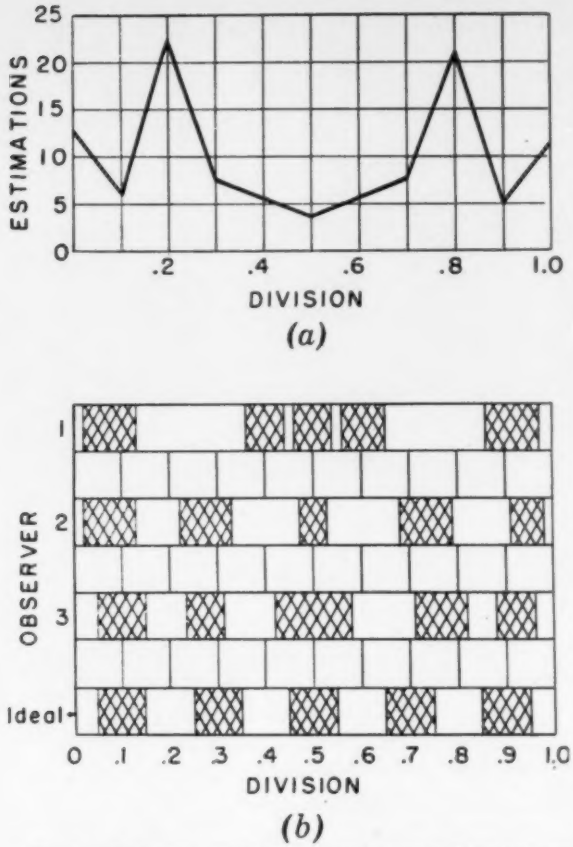


FIG. 3. a. Subjective estimations of meter pointer positions between divisional marks.
b. Typical scatter of subjective data.

sharply as the division width increased to 2 mm and more. Fig. 5 illustrates the results. From this accumulation of data, Harris(7) concludes that as a general case, observations by several individuals would be necessary to ensure that an estimation of pointer position was correct within a one-tenth division.

TYPICAL DEVIATIONS FROM ACCURACY STANDARDS

The foregoing has been offered as a preliminary briefing on the importance of standards of accuracy in measurements employing deflection instruments. It is far from complete but should serve to alert the laboratory worker to the danger of overrating the ability of instruments to give accurate measurements.

To demonstrate the practicality of this discussion, it is pertinent to touch briefly on personal experiences with measurements which are appropriate to a consideration of accuracy, as well as precision.

In the determination of response of the human ear, the exploration of acoustical properties of enclosures or materials, the study of sound transmission or reflection under water, and many other audio research problems, extensive use is made of peaked or pulse-type signals. The magnitude of these signals is determined principally by physical measurements of cathode ray presentations or by direct reading of deflection instruments. Since the former method is hardly suited for accurate measurements, attention is directed toward the second method, or the use of deflection instruments.

Measurements made in studies of psychophysical responses are subject to disturbances in repeatability which serve as a continual source of worry to a worker in the laboratory. Recourse is had to multiplicity of similar measurements so that by statistical means deviations may be determined and proper evaluation made of the detrimental effect.

A typical data curve is one such as Fig. 6, which indicates response of a commercial earphone when employed with a number of subjects. Deviations were computed to account for the differential in hearing of the several subjects. But it should be noted that at the level of measurement made, instrument error of $\pm$ 3% of reading is greater than most of the deviations shown!

Another example is in the determination of cochlear potentials of animals under various stimulation or depressive controls. Such work, of an extremely delicate nature, requires recording tof potentials in the microvolt region. No attempt will be made here to comment on the effect of physiological noise upon the sought-for signal. More important is to bear in mind that instrumentation commonly in use has an overall error of $\pm$ 5%. Consequently, application of data presented in the form of smoothed curves should be made with discretion. Standard deviations to allow for subject differentials ought to be reconsidered and recalculated to take into consideration instrument error, and these deviations prominently displayed on such curves.

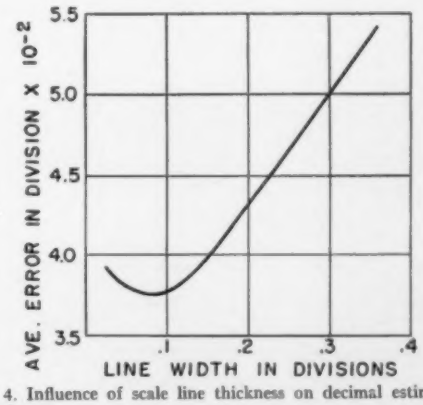


FIG. 4. Influence of scale line thickness on decimal estimation of meter pointer.

An assembly of instruments consisting of condenser microphone, preamplifier, band pass filter, and electronic voltmeter is employed in psychoacoustic laboratories for sound-pressure measurements under various conditions. The assembly normally is employed to give measurements believed to be correct to ± 0.2 db. Calibration of the entire system is not difficult or lengthy. However, this writer has observed the assembly in use under conditions which void the accuracy statements of the manufacturer. These were observation of complex signals, detection of flutter-modulated signals, and measurement of high intensity levels. The manufacturer states that the electronic meter's averaging characteristics are within 1.0 db of square law for complex waves —an inherent error when signals other than sinusoidal are being observed. The manufacturer states that the meter's damping provides a dynamic characteristic which is neither too fast nor too slow, furnishing *reasonable* electrical averaging of rapid changes—an ambiguous specification for measurement accuracy when one is studying flutter effects. The manufacturer states measurements within the range of the instrument can be made to an accuracy of 0.2 db; assuming a *steady* sound level. The manufacturer states that indicators are provided to assure the operator that the electrical level at each position of the circuit is properly adjusted to avoid spurious readings due to overload. However, these indicators are available to the operator only after operation of a switching circuit, a function normally not employed by operators who are unaware of the special precautionary statements in the manufacturer's specifications.

The foregoing is not to be construed as criticism of this particular unit since the writer has employed it satisfactorily under controlled conditions for several years. It should be construed as a definite hint that laboratory workers can never know too much about their equipment and even when familiar with units should occasionally review the manufacturer's claims and statements of limitations of guaranteed accuracy.

In conclusion, it is of the greatest importance to every laboratory worker to bear in mind the principle that uncertainty in measurement accuracy is inherent. A knowledge of errors should be developed and an understanding of limitations promoted. With such a background, then, he may announce to the world, "Our research has proven that our presumptions were correct" and add with humility, "to the best of our measurement ability."

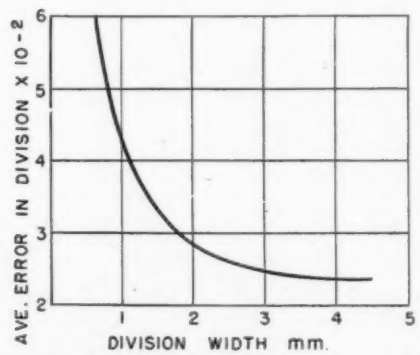


FIG. 5. Influence of scale division width on accuracy of estimation of meter pointer.

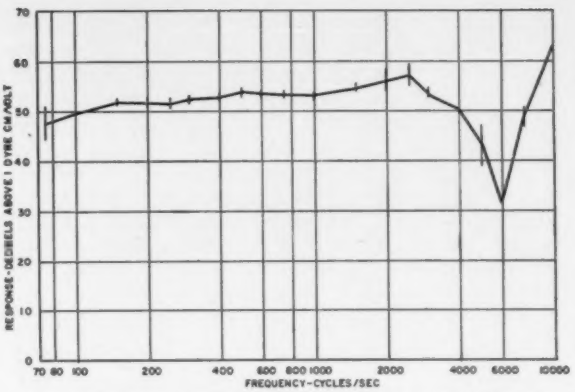


FIG. 6. Response of commercial earphone and subject deviation.

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THE AUTHOR

Charles E. White studied electrical engineering while he was employed in Western Union Research Laboratories, receiving his B.S. degree from New York University in 1934. He later completed work for the M.S. at the University of Connecticut. During his professional career, including U. S. Navy service 1942-1946, he has been concerned with electronics and acoustics problems, conducting both basic and applied research. At AVCO R.A.D., which he joined in 1956, he has dealt chiefly with coordination of laboratory standards. He holds memberships in IRE, ASA, AES, SMPTE, the International Radio Scientific Union, and the U. S. National Committee on Radio Frequency Standards.

AES News

First Report of the AES Technical Committee on Magnetic Recording

Creation of the Committee

As announced in the January, 1959, JOURNAL AES, the Board of Governors of the Audio Engineering Society has approved the creation of an AES Technical Committee on Magnetic Recording with the undersigned as Chairman. The Board approved the following scope of the Committee's work:

To investigate new products, new techniques, and research in the field of magnetic recording; to analyze their application to the audio engineering field; and to report to the Society in the JOURNAL OF THE AUDIO ENGINEERING SOCIETY.

The scope of the committee's activities does not include magnetic recording standards; however, the committee may report on the standardization activities of other committees.

The Committee's Activities

This Committee can provide a useful service to the magnetic recording industry only if those working in the field of magnetic recording contribute to its activities. This contribution can be made by submitting for publication in the JOURNAL any material that may be of interest to audio engineers working in the field of magnetic recording. The Board of Governors has outlined the scope of the committee's activities in the opening paragraph of this report as a guide for subject matter. The statement of the Board refers to new products and new techniques. However, there are many useful items of engineering information buried in old reports and notebooks that would be new to most of us simply because they have never been published. The reports of the Committee to be published in the JOURNAL will provide an outlet for these items that might not be considered suitable material for a formal paper. The length of an item to be included in the report could be as short as a photograph or a graph with a single-line caption, or as long as one page of the JOURNAL, equivalent to about 1000 words. The use of appropriate illustrations is encouraged; however, they should consist of glossy photographic prints, ink drawings, or photostat copies of ink drawings. Those submitting material will be given specific credit by their name and company affiliation for each item included as part of the report of the Committee. Everyone working in the field of magnetic recording is invited to submit any report material within the outlined scope of the Committee to the Chairman at the address given at the end of this report. An example of the type of material that will be included in future reports is as follows:

Low-Distortion German Tape

A German iron-oxide coated tape made over ten years ago was found to have a most exceptional non-linear distortion characteristic, as shown in the table below:

TYPE LG GERMAN TAPE
400 cps 15 ips

Playback output level, db below saturation	Percent harmonic distortion	
	Third	Fifth
0	4.4	1.7
1	1.3	0.4
2	0.7	0.6
3	0.7	0.6
6	1.2	0.4
12	0.8	—

Second harmonic distortion below 0.1% at all levels.

This particular sample of Type LG tape has much lower distortion near saturation than any of the other LG tapes tested. It definitely is not a typical sample of LG tape. This tape is believed to be unique among all magnetic recording media in that the total harmonic distortion is less than 1.5% for all playback output levels one db or more below saturated output. The distortion characteristic is very similar to that of an audio amplifier with a moderate amount of negative feedback. Unfortunately the identification of this particular sample of LG tape is not sufficiently detailed to be of any value in locating this roll of tape, if indeed it still exists.

Chairman of the AES Technical Committee on Magnetic Recording.

WALTER H. ERIKSON
Radio Corporation of America
Building 10-8
Camden 2, New Jersey

Award to AES Member and Author

The Board of the Scientific Radio Fund "Veder" have awarded a prize to Mr. R. Vermeulen, scientific adviser to the Philips Research Laboratory and head of the Acoustical Group. The Veder Fund was established by Mr. Veder of Rotterdam, one of the founders and promoters of the Dutch Association of Radio Engineering (NVVR).

Dealing with the field of spatial sound reproduction Mr. Vermeulen, in cooperation with his associate Mr. Kleis, developed a system for improving the acoustical properties of a room, the artificial reverberation system. The equipment used renders it possible to "surround" the auditor with sound, so to speak, in the same way as this can be perceived in a concert hall with good acoustical properties. The artificial reverberation was publicly demonstrated for the first time during the International Acoustical Congress in 1953. After that the equipment was used, to mention a few instances in the Hall of Arts and Sciences at The Hague, the world-famous Teatro alla Scala at Milan, the Palais de Chaillot in Paris and the Grand Auditorium of the Brussels Universal Exhibition 1958.

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(Fall Convention, 1958 to April 1, 1959)

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STUDENT SECTION NOTES

The Student Section at Indiana Technical College reported at its January meeting the results of elections. Mr. Gordon Honert is Chairman, with Mr. Willard Jones in the post of Vice Chairman. Mr. Gene Faulkenberry is Secretary and Mr. Earl Kepner is Treasurer. The Executive Committee consists of Messrs. James Marello and Garland Linde. In February, a film entitled "Transistors" was presented.

A February meeting of Rensselaer Student Section presented Prof. G. F. Robinson, Head of RPI Acoustics Laboratories, in a lecture "Use of Analogue Computers in Audio Transducer Design." The membership, composed of both upper and lower classmen, found the presentation most informative.

MISCELLANEOUS NEWS AND INFORMATION

International Electrotechnical Commission to Meet in Madrid—About fifty representatives from the United States are expected to attend the 1959 meeting of the International Electrotechnical Commission (IEC), to be held in Madrid, Spain, June 30 to July 10. Representatives from thirty-three countries are expected to participate in the work of thirty technical committees. Other IEC technical committees will meet this year in Paris and London. Founded in 1904, the IEC is active in the development of international standards in the electrotechnical and related fields.

IEC Booklet Available—A new thirty-page booklet describing the International Electrotechnical Commission (IEC)—what it is; what it does; and how it works—is available at no cost from the American Standards Association.

IRE National Convention—New York, March 23 through 26, was the scene of the IRE National Convention. At the annual banquet on March 25, Lloyd V. Berkner, president of Associated Universities, Inc., and international president of the International Scientific Radio Union, was the principal speaker; his topic, "The IRE Enters Space." 950 exhibitors took up all four floors of the Coliseum and provided visitors with the most complete showcase of new apparatus and products ever assembled under one roof.

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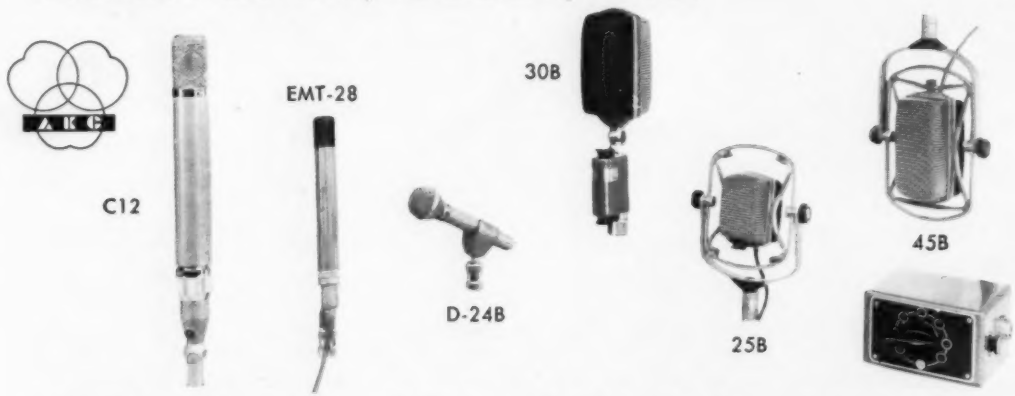

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REK-O-KUT COMPANY